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IN REPLY REFER TO: Joint Interoperability Test Command (JTE) **31 Mar 10**

MEMORANDUM FOR DISTRIBUTION

SUBJECT: Special Interoperability Test Certification of Cisco Unified Communication Manager Version 7.1(2) with Internetwork Operating System (IOS) Software Release 12.4(22)T2

References: (a) DoD Directive 4630.5, "Interoperability and Supportability of Information Technology (IT) and National Security Systems (NSS)," 5 May 2004
(b) CJCSI 6212.01D, "Interoperability and Supportability of Information Technology and National Security Systems," 8 March 2006
(c) through (f), see Enclosure 1

1. References (a) and (b) establish the Defense Information Systems Agency (DISA), Joint Interoperability Test Command (JITC), as the responsible organization for interoperability test certification.

2. The Cisco Unified Communications Manager Version 7.1(2) with IOS Software Release 12.4(22)T2 is hereinafter referred to as the system under test (SUT). The SUT meets all of the critical interoperability requirements and is certified for joint use within the Defense Switched Network (DSN) for the following switch types: Private Branch Exchange (PBX) 1 and PBX 2. The SUT meets the Voice over Internet Protocol (VoIP) critical interoperability requirements with any certified Assured Services Local Area Network (ASLAN) or ASLAN components on the Unified Capabilities (UC) Approved Products List (APL). The identified test discrepancies shown in the Certification Testing Summary (Enclosure 2) have an overall minor operational impact. No other configurations, features, or functions, except those cited within this report, are certified by the JITC. This certification expires upon changes that could affect interoperability, but no later than three years from the date of this memorandum.

3. This finding is based on interoperability testing conducted by JITC, DISA adjudication of open test discrepancy reports, review of the vendor's Letters of Compliance (LoC), and Defense Information Assurance (IA)/Security Accreditation Working Group (DSAWG) accreditation. Interoperability testing of the SUT was conducted at JITC's Global Information Grid Network Test Facility at Fort Huachuca, Arizona, from 14 July through 22 August 2009. Additional testing was conducted from 9 through 20 November 2009. DISA adjudication of outstanding test discrepancy reports was completed on 2 September 2009. Review of the vendor's LoC was completed on 29 September 2009. DSAWG granted accreditation on 10 March 2010 based on the security testing completed by DISA-led IA test teams and published in a separate report,

Reference (c). Enclosure 2 documents the test results and describes the tested network and system configurations.

4. The SUT certified hardware and software components are listed in Table 1. The interoperability test summary of the SUT is indicated in Table 2. The PBX 1 Capability Requirements (CRs) and Feature Requirements (FRs) are listed in Table 3. This interoperability test status is based on the SUT's ability to meet:

a. DSN services for Network and Applications specified in Reference (d).

b. PBX 1 interface and signaling requirements for trunks/lines specified in Reference (e) verified through JITC testing and/or vendor submission of LoC.

c. PBX 1 CRs/FRs specified in Reference (e) verified through JITC testing and/or vendor submission of LoC.

d. The overall system interoperability performance derived from test procedures listed in Reference (f).

e. The IPv6 requirements specified in References (e) and (g).

f. The softphone requirements specified in Reference (h).

Table 1. SUT Hardware and Software Components

Cisco Unified Communications Manager Version 7.1(2), with IOS Software Release 12.4(22)T2			
Component (See note 1.)	**Release**	**Sub-component (See note 1.)**	**Function**
Communication Managers **<u>MCS7835I2, MCS7835H2</u>, <u>MCS7825H3</u>, <u>MCS7825H4</u>,** MCS7835H1, MCS7845H1, MCS7845H2, MCS7825I4, MCS7835I1, MCS7845I1, MCS7845I2	7.1(2.30005.2)	Not Applicable	Processing/Signaling
<u>Cisco 3845</u>, 3825 Integrated Services Router (Gateway)	IOS 12.4(22) T2	**<u>NM HDV2</u>**	TDM Interface NM, HD Voice, 2-slot IP communications enhanced voice/fax
		<u>VWIC2 2MFT T1/E1</u>	Second Generation Voice/WAN Interface Card 2-port RJ-48, Multiflex Trunk T1/E1 (See note 2.)
		<u>NM HDV2 2T1/E1</u>	2-port T1/E1 IP Communications HD voice/fax NM 2 T1/E1 controllers (See note 2.)
		<u>NM HDV2 1T1/E1</u>	1-port T1/E1 IP Communications HD voice/fax NM 2 T1/E1 controllers (See note 2.)
		<u>VIC3 4FXS/DID</u>	Voice Interface Card, 4-port, RJ-11, Foreign Exchange Station, DID
		<u>VIC3 2FXS</u>	Voice Interface Card, 2-port, Foreign Exchange Station
		<u>VIC 4FXS/DID</u>	Voice Interface Card, 4-port, RJ-11, Foreign Exchange Station, DID
		VIC2 2FXS	Voice Interface Card, 2-port, RJ-11, Foreign Exchange Station
		<u>EM HDA 8FXS</u>	8-port analog Foreign Exchange Station expansion module for voice and fax (See note 3.)
		<u>EVM HD 8FXS/DID</u>	HD analog and digital extension module for voice and fax
		<u>EM3 HDA 8FXS/DID</u>	8-Port HD analog and digital extension module for voice and fax (See note 3.)

Table 1. SUT Hardware and Software Components (continued)

Cisco Unified Communications Manager Version 7.1(2) with IOS Software Release 12.4(22)T2 (continued)			
Component (See note 1.)	**Release**	**Sub-component (See note 1.)**	**Function**
Cisco 2851, 2821, 2811 Integrated Services Router (Gateway)	IOS 12.4(22)T2	NM HD 2VE	2-slot IP communications enhanced voice/fax network module
		VIC 4FXS/DID	Voice Interface Card, 4-port, RJ-11, Foreign Exchange Station, DID
		VWIC2 2MFT T1/E1	Second Generation Voice/WAN Interface Card 2-port RJ-48, Multiflex Trunk T1/E1 (See note 2.)
		EVM HD 8FXS/DID	HD analog and digital extension module for voice and fax
		EM HDA 8FXS	8-port analog Foreign Exchange Station expansion module for voice and fax (See note 3.)
		NM HDV2 2T1/E1	2-port T1/E1 IP Communications HD voice/fax NM, 2 T1/E1 controllers (See note 2.)
		NM HDV2 1T1/E1	1-port T1/E1 IP Communications HD voice/fax NM, 1 T1/E1 controllers (See note 3.)
		VWIC2 1MFT T1/E1	Second Generation Voice/WAN Interface Card 1-port RJ-48, Multiflex Trunk T1/E1 (See note 2.)
		VIC3 4FXS/DID	Voice interface card, 4-port, RJ-11, foreign exchange station, DID
		EM3 HDA 8FXS/DID	8-port HD analog and digital extension module for voice and fax (See note 3.)
		VIC3 2FXS	Voice Interface card, 2-port, RJ-11, Foreign exchange station
		VIC 4FXS/DID	Voice Interface Card, 4-port, RJ-11, Foreign Exchange Station, DID
		VIC2 2FXS	Voice Interface Card, 2-port, RJ-11, Foreign Exchange Station
CP-7940G and CP-7960G (See note 4.)	P00308010100	Not Applicable	IP Phone (with push-to-talk handset or with standard handset)
CP-7970G and CP-7971G	SCCP70.8-5-2S	Not Applicable	IP Phone (with push-to-talk handset or with standard handset)
CP-7931G	SCCP31.8-5-2S	Not Applicable	IP Phone (with push to talk handset or with standard handset)
CP-7911G and 7906G	SCCP11.8-5-2S	Not Applicable	IP Phone (with push-to-talk handset or with standard handset)
CP-7941G, CP-7941G-GE, CP-7961G, and CP-7961G-GE	SCCP41.8-5-2S	Not Applicable	IP Phone (with push-to-talk handset or with standard handset)
CP-7942G and CP-7962G	SCCP42.8-5-2S	Not Applicable	IP Phone (with push-to-talk handset or with standard handset)
CP-7945G and CP-7965G	SCCP45.8-5-2S	Not Applicable	IP Phone (with push-to-talk handset or with standard handset)
CP-7975G	SCCP75.8-5-2S	Not Applicable	IP Phone (with push-to-talk handset or with standard handset)
7914	Load: S00105000400	Not Applicable	Expansion module
7915	B015-1-0-3	Not Applicable	Expansion module
7916	B015-1-0-3	Not Applicable	Expansion module
General Dynamics C4 Systems Sectéra® vIPer™ (See note 5.)	Release 1.0, Software ver.6.04	Not Applicable	IP Phone (with standard handset)
Telecore 2151	2AE-00056-0003	Not Applicable	IP Phone (with push-to-talk handset or with standard handset), 100 Mbps shared access[6]
CIS Secure DTD-7961-T-SG-SC-SC-X-X (See note 7.)	SCCP41.8-5-2S	Not Applicable	7961G TEMPEST version with 100 Mbps SC Fiber LAN and PC interfaces, TSG Positive Disconnect, no speakerphone, shared access
CIS Secure DTD-7975-X-XSC-RJ-ME-SE (See note 7.)	SCCP75.8-5-2S	Not Applicable	7975G Standard with 1000 Mbps SC Fiber LAN and RJ45 PC interfaces, shared access
CRYPTEK CT915-V-P1-003 (See note 7.)	SCCP41.8-5-2S	Not Applicable	7961G IP phone, Fiber TEMPEST version with 100MB Fiber LAN and no shared access

Table 1. SUT Hardware and Software Components (continued)

Cisco Unified Communications Manager Version 7.1(2) with IOS Software Release 12.4(22)T2 (continued)			
Component (See note 1.)	**Release**	**Sub-component (See note 1.)**	**Function**
Walker WS-2620	Not Applicable	Not Applicable	Push to Talk Handset for Cisco 7900 Series phones
Cisco IP Communicator (See note 8.)	7.0.5	Not Applicable	Cisco Softphone Application

NOTES:

1 Components bolded and underlined were tested by JITC. The other components in the family series were not tested; however, they utilize the same software and similar hardware and JITC analysis determined them to be functionally identical for interoperability certification purposes and they are also certified for joint use.

2 These components are certified in the DSN with T1 ISDN PRI interface. These components are certified in the PSTN with the T1 ISDN PRI and E1 ISDN PRI interfaces.

3 The EM HDA 8FXS and EM3 HDA 8FXS/DID expansion modules require the EVM HD module. Up to two EM HDA 8FXS or EM3 HDA 8FXS/DID expansion modules are supported for each EVM HD.

4 The Cisco CP-7940G and CP-7960G end instruments did not meet dual stack IPv6 requirements. These end instruments represent legacy end instruments which are IPv4 only; however, the SUT met the minimum requirement for dual stack IPv6 end instruments with the other IP end instruments listed in this table and a dual stack call control agent in accordance with the interim UCR IPv6 rules of engagement, Reference (g).

5 This instrument is certified specifically with 2800 and 3800 series gateways with IOS 12.4(22) T2 or higher version listed on the UC APL.

6 Although the Telecore 2151 supports both 100 Mbps and 1 Gbps shared access, due to MOS scores below the required 4.0 for 1 Gbps shared access, the Telecore 2151 is only certified for shared access at 100 Mbps.

7 CIS Secure Computing and Cryptek add security modifications which are physical in nature and do not affect software. The CIS Secure Computing and Cryptek security modifications can be applied to any certified Cisco IP phone.

8 Reference (h) is a DISA memo that stipulates interim softphone requirements that supersede the current UCR 2008 requirements until they are implemented in Change 1. The softphone shall be functionally identical to a traditional IP "Hard" telephone and will be required to provide voice features and functionality provided by a traditional IP "Hard" Telephone with following exceptions:
   a. Audible and visual alerting to the end user of an incoming call, even if the application is running in the background.
   b. Softphone application shall be exempt from reliability, availability and performance (packet loss, jitter, latency) requirements.
   c. Microphone and speaker or headphone, or any other audio input/output device, Ethernet interface(s), and mouse (point and click) interaction.
   d. IPv6 is not required.

LEGEND:

APL	Approved Product List
CP	Cisco Phone
DID	Direct Inward Dialing
DISA	Defense Information Systems Agency
DSN	Defense Switched Network
E1	European Basic Multiplex Rate (2.048 Mbps)
EM	Expansion Module
EVM	Extension Voice Module
Fax	facsimile
FXS	Foreign Exchange Station
Gbps	Gigabits per second
G	10/100BaseT Ethernet (A Cisco part designator on their IP phone.)
GE	Gigabit Ethernet (A Cisco part designator on their IP phone.)
HD	High Density
HDA	High Density Analog
IOS	Internetwork Operating System
IP	Internet Protocol
Ipv4	Internet Protocol version 4
IPv6	Internet Protocol version 6
ISDN	Integrated Services Digital Network
JITC	Joint Interoperability Test Command
LAN	Local Area Network
Mbps	Megabits per second
MCS	Media Convergence Server
MFT	Multiflex Trunk
MOS	Mean Opinion Score
NM	Network Module
PC	Personal Computer
PRI	Primary Rate Interface
PSTN	Public Switched Telephone Network
RJ	Registered Jack
SC	fiber connector (square push-in)
SCCP	Skinny Call Control Protocol
SUT	System Under Test
T1	Digital Transmission Link Level 1 (1.544 Mbps)
TDM	Time Division Multiplexing
UC	Unified Capabilities
UCR	Unified Capabilities Requirements
V	Voice
VE	Voice/Fax Enhanced
VIC	Voice Interface Card
VWIC	Voice WAN Interface Card
WAN	Wide Area Network

Table 2. SUT Interoperability Test Summary

DSN Trunk Interfaces			
Interface & Signaling	**Critical**	**Status**	**Remarks**
T1 CAS (DTMF, MFR1, DP)	No	Not Certified	The SUT T1 CAS interface was tested but did not meet all critical CRs and FRs. The SUT T1 CAS interface is therefore not certified by JITC. This is not a required interface for a PBX 1.[1]
E1 CAS (DTMF, MFR1, DP)	No (Europe only)	Not Tested	E1 CAS is supported by the SUT. However, it was not tested. The SUT E1 CAS interface is therefore not certified by JITC. This is not a required interface for a PBX 1.
T1 ISDN PRI NI 1/2 (ANSI T1.619a)	Yes	Certified	Met all critical CRs and FRs with the following minor exception: The SUT T1 ISDN PRI NI2 interface does not support NFAS.[2]
E1 ISDN PRI (ITU-T Q.955.3)	No (Europe only)	Not Certified	The E1 ISDN PRI interface is supported by the SUT; however, it does not support ITU-T Q.955.3 MLPP. The SUT E1 ISDN PRI interface is therefore not certified by JITC for use within the DSN. This interface is certified only for PSTN. This is not a required DSN interface for a PBX 1.
DSN Line Interfaces			
Interface & Signaling	**Critical**	**Status**	**Remarks**
2-Wire Analog Loop Start (GR-506-CORE)	Yes	Certified	Met all critical CRs and FRs with the following minor exception: The SUT gateway analog interface does not support required line features.[3] The operational impact is minor.
ISDN BRI NI 1/2 (ANSI T1.619a)	No	Not Tested	This interface is not supported by the SUT. This is not a required interface for a PBX 1. There is no risk associated with the SUT not supporting this interface.
2-Wire Proprietary Digital	No	Not Tested	This interface is not supported by the SUT. This is not a required interface for a PBX 1. There is no risk associated with the SUT not supporting this interface.
VoIP (Ethernet IEEE 802.3u)	No	Certified	Met all critical CRs and FRs with the following minor exception: The Cisco CP-7940G and CP-7960G end instruments did not meet dual stack IPv6 requirements.[4]
DSN Features and Capabilities			
Features and Capabilities	**Critical**	**Status**	**Remarks**
Common Features	Yes	Certified	Met all critical CRs and FRs with the following minor exception: Full compliance of DSN Common Call Features was not met. The operational impact is minor.[3]
Attendant	No	Not Tested	This feature is not supported by the SUT. This is not a required feature for a PBX 1. There is no risk associated with the SUT not supporting this feature.
Public Safety	Yes	Certified	All public safety features are conditional. The SUT met all critical CRs and FRs for Basic 911. The SUT does not support the other public safety features. These are not required features for a PBX 1. There is no risk associated with the SUT not supporting these features.[5]
Conferencing	No	Not Certified	The SUT can support Meet-Me Conferencing through the optional MeetingPlace Express.[6] The SUT does not support Preset Conferencing or Progressive Conferencing. These features are not required for a PBX 1. There is no risk associated with the SUT not supporting these features.
Nailed-up Connections	No	Not Tested	This feature is not supported by the SUT. This is not a required feature for a PBX 1. There is no risk associated with the SUT not supporting this feature.
DSN Hotline Services	No	Not Tested	This feature is not supported by the SUT. This is not a required feature for a PBX 1. There is no risk associated with the SUT not supporting this feature.
MLPP	Yes	Certified	Met all critical CRs and FRs with the following minor exceptions: The SUT does not support a global diversion number.[7] The SUT does not support the Loss of Command and Control announcement.[8]
Call Processing	Yes	Certified	Met all critical CRs and FRs.
ISDN Services	Yes	Not Tested	This feature is not supported by the SUT. This is not a required feature for a PBX 1. There is no risk associated with the SUT not supporting this feature.
Synchronization	Yes	Certified	Met all critical CRs and FRs.
Reliability	Yes	Certified	Met all critical CRs and FRs.
Security	Yes	Certified	See note 9.
VoIP System	No	Certified	The SUT is certified for VoIP specifically with any certified ASLAN or ASLAN components posted on the UC APL. (See notes 4 and 10.)

Table 2. SUT Interoperability Test Summary (continued)

Network Gateways				
Gateway	**Interface & Signaling**	**Critical**	**Status**	**Remarks**
PSTN	T1 CAS (DTMF, MFR1, DP)	No	Not Certified	SUT T1 CAS interface was tested but did not meet requirements. The SUT T1 CAS interface is therefore not certified by JITC. This is not a required interface for a PBX 1.[1]
	E1 CAS (DTMF, MFR1, DP)	No (Europe only)	Not Tested	E1 CAS is supported by the SUT; however it was not tested. The SUT E1 CAS interface is therefore not certified by JITC. This is not a required interface for a PBX 1.
	T1 ISDN PRI NI 1/2 (ANSI T1.607)	No	Certified	Met all critical CRs and FRs with the following minor exception: The SUT T1 ISDN PRI NI/2 interface does not support NFAS.[2]
	E1 ISDN PRI (ITU-T Q.931)	No (Europe only)	Certified	Met all critical CRs and FRs.
	2-Wire Analog Ground Start (GR-506-CORE)	No	Certified	Met all critical CRs and FRs.[11]

NOTES:

1 The SUT T1 CAS interface does not recognize Remove from Service (Busy Out) or Restore to Service (Make Idle) condition from the distant end switch. In addition, when the Busy Out condition is invoked across the T1 CAS interface, it causes the SUT 3845 and 2851gateway T1 CAS interface to deregister from its current subscriber and reregister to an alternate subscriber and then within 1 to 5 minutes repeat the process and go back to its original subscriber. During this transition period, calls are unable to process to the SUT.

2 The SUT does not support NFAS on their ISDN PRI NI2 interface. DISA's adjudication of this discrepancy was completed on 17 December 2008 and was ruled to have a minor operational impact. Furthermore, DISA stated they intent to modify the next update of the UCR to change NFAS for a PBX 1 from required to conditional.

3 All of the features on the VoIP phones were tested using multiple line appearances. A minimum of two line appearances is required to meet the MLPP interoperability requirements for Call Features with the exception of call hold, call pickup, and call forwarding functions. Call Forward No Answer, Call Forward Busy, and Multi-Line Hunt Service are supported on both VoIP and analog phones. Call Forward Variable, Three-way Calling, Call Hold, Call Pick-up, and Call Transfer are supported on VoIP phones only. These features are required for a PBX 1 for all instruments, however since this requirement is a new UCR requirement and the vendor has 18 months to develop it (July 2010), the operational impact is minor. Since the SUT test window started before the 18 month development window expired, DISA stated this new feature requirement does not apply. All of the features on the VoIP phones were tested using multiple line appearances. A minimum of two line appearances is required to meet the MLPP interoperability requirements for Call Features with the exception of call hold, call pickup, and call forwarding functions. Although the SUT does not support Precedence Call Waiting, they do support multiple call appearances on their VoIP phones. This provides the ability for a user to receive additional calls while active with another call. A short "ping" ring is not provided when calls are forwarded; however, the phone does visually display that call forward variable is enabled. There is a minor operational impact. When a ROUTINE call is placed to a hunt group, and a ring-no-answer condition occurs, the calling party is diverted to the MLPP alternate directory number. This configuration must be done to allow correct treatment to be provided to precedence calls above ROUTINE that are placed to the hunt group. There is no operational impact.

4 The SUT met all IPv6 requirements through testing and LoC with the following exception: The Cisco CP-7940G and CP-7960G end instruments did not meet dual stack IPv6 requirements. These end instruments represent legacy end instruments which are IPv4 only; however, the SUT met the minimum requirement for dual stack IPv6 end instruments with the other IP end instruments listed in this table and a dual stack call control agent in accordance with Reference (g).

5 The SUT only supports emergency service 911 public safety features. The following public safety features are not supported and therefore are not covered in this certification: Trace of terminating calls, Outgoing call trace, Tandem call trace, and Trace of a call in progress. There is no operational impact because these public safety features are not required for a PBX 1.

6 Meet-Me Conferencing can be met through the use of an optional adjunct conferencing system called the Cisco Meeting Place Express which is covered under a separate certification.

7 The SUT does not support an MLPP global diversion number. Each station must be individually configured with a precedence diversion number from a single location using the Bulk Administration Tool provided with the Cisco Unified Communication Manager. The operational impact is minor because diversion settings can be configured for all of the stations provisioned on the switch from a single location.

8 The SUT does not support the Loss of C2 announcement. This announcement is invoked only when a DSN subscriber is automatically routed to a non-MLPP network. DISA adjudicated this anomaly as having a minor operational impact because this announcement would rarely be invoked on a PBX 1. Furthermore, DISA, in coordination with the Joint Staff, stated their intent to modify the next update of the UCR to change the Loss of C2 announcement from required to conditional for a PBX 1.

9 Security is tested by DISA-led Information Assurance test teams and published in a separate report, Reference (c).

Table 2. SUT Interoperability Test Summary (continued)

NOTES continued:

10 The following discrepancies noted with the SUT were adjudicated by DISA on 2 September 2009 as having a minor operational impact:

a. The VoIP SUT session control components and end instruments can only have the signaling service Traffic Class configured for 21 different DSCP values and not the full range required of 0-63.
b. The MCS7835 and the MCS7825 call managers OAM traffic is tagged at zero and is not configurable.
c. The 2851 and 3845 gateways are tagging IPv4 RTCP traffic at zero and it is not configurable.
d. When the CP-7940G and CP-7960G phones are powered up, some of the UDP/TFTP traffic has a DSCP value of 4 and 802.1Q value of 5 and can not be changed.
e. The SUT management workstation provided during testing did not assign DSCP values for OAM IP traffic.
f. IP phones are incorrectly tagging IPv6 TCP traffic during power up.
g. Soft Client is incorrectly tagging all traffic during power up.
h. The 802.1Q COS tag values are not independently configurable from the DSCP values.
i. The MCS7825H4 Communication Manager server stopped transmitting IP Traffic. The NIC failover must be disabled to correct this problem. The NIC failover is offered on this server but is not required for a PBX1. NIC failover is not certified for any server platform and should not be enabled. This setting will be annotated in the deployment guide for this server.
j. End Instruments, except for the Telecore 2151, do not support the manual configuration of the IPv6 default gateway.
k. Communication Managers are incorrectly tagging UDP/TFTP traffic to the end instrument during end instrument power up.

11 This interface requirement was met by the vendor's LoC.

LEGEND:

802.1Q	Standards for Local and Metropolitan Area Networks: Virtual Bridged Local Area Networks	LoC	Letters of Compliance
		LSSGR	Local Access and Transport Area (LATA) Switching Systems Generic Requirements
802.3u	Standard for carrier sense multiple access with collision detection at 100 Mbps	Mbps	Megabits per second
		MCS	Media Convergence Servers
ANSI	American National Standards Institute	MFR1	Multi-Frequency Recommendation 1
APL	Approved Products List	MLPP	Multi-Level Precedence and Preemption
ASLAN	Assured Services Local Area Network	ms	milliseconds
BRI	Basic Rate Interface	NI 1/2	National ISDN Standard 1 or 2
C2	Command and Control	NI2	National ISDN Standard 2
CAS	Channel Associated Signaling	NIC	Network Interface Card
CoS	Class of Service	NFAS	Non Facility Associated Signaling
CP	Cisco Phone	OAM	Operational Administration and Maintenance
CRs	Capability Requirements	PBX 1	Private Branch Exchange 1
DISA	Defense Information Systems Agency	PMO	Program Management Office
DP	Dial Pulse	PNT	Preemption Notification Tone
DSCP	Differentiated Services Code Point	PRI	Primary Rate Interface
DSN	Defense Switched Network	PSTN	Public Switched Telephone Network
DSS1	Digital Subscriber Signaling 1	Q.931	Signaling Standard for ISDN
DTMF	Dual Tone Multi-Frequency	Q.955.3	ISDN Signaling standard for E1 MLPP
E1	European Basic Multiplex Rate (2.048 Mbps)	RTCP	RTP Control Protocol
EI	End Instrument	RTP	Real-time Transport Protocol
FRs	Feature Requirements	SS7	Signaling System 7
GR	Generic Requirement	SUT	System Under Test
GR-506-CORE	LSSGR: Signaling for Analog Interfaces	T1	Digital Transmission Link Level 1 (1.544 Mbps)
ICA	Isolated Code Announcement	T1.607	ISDN Layer 3 Signaling Specification for Circuit Switched Bearer Service for DSS1
IEEE	Institute of Electrical and Electronics Engineers		
IP	Internet Protocol	T1.619a	SS7 and ISDN MLPP Signaling Standard for T1
IPv4	Internet Protocol version 4	TCP	Transmission Control Protocol
IPv6	Internet Protocol version 6	TFTP	Trivial File Transfer Protocol
ISDN	Integrated Services Digital Network	UC	Unified Capabilities
ITU-T	International Telecommunication Union - Telecommunication Standardization Sector	UCR	Unified Capabilities Requirements
		UDP	User Datagram Protocol
JITC	Joint Interoperability Test Command	VoIP	Voice over Internet Protocol

Table 3. PBX 1 Requirements

DSN Trunk Interfaces				
Interface	**Critical**	**Requirements Required or Conditional**		**References**
T1 CAS (MFR1, DTMF, DP) E1 CAS (MFR1, DTMF, DP) T1 ISDN PRI NI 1/2 (ANSI T1.619a) E1 ISDN PRI (ITU-T Q.955.3)	No No (Europe only) Yes No (Europe only)	Trunking	• PBX Line (C) • Direct Inward Dialing (C) • National ISDN 1/2 Primary Access (R) • ISDN ANSI MLPP Service Capability (R) • ITU-T ISDN Primary Access (Europe only) (C) • ITU-T ISDN Primary Access Digital Subscriber Signaling System Number 1 MLPP (Europe only) (C) • Normal Wink Start Operations (R) • Glare Operation (R) • Abnormal Wink Start (R) • Glare Resolution (R) • Call for Service Timing (R) • Guard Timing (R) • Satellite Timing (R) • Disconnect Control (R) • Reselect and Retrial (R) • Off-Hook Supervision Transition (R) • Dial-Pulse Signals (R) • DTMF Signaling (R) • Standard Digit Format for Precedence (C) • MFR1 2/6 Signaling (C) • Alerting Signals and Tones (R) • DSN ISDN User-to-Network Signaling (R) • Application (R) • Physical Layer (R) • Data Link Layer (R) • Data Link Connection (R) • Peer-to-Peer Procedures of Data-Link Layer (R) • Layer 3 DSN User-to-Network Signaling (R) • DSN User-to-Network Signaling for Circuit-Switched Bearer Services (R) • Sequence of Messages for DSN Circuit-Switched Calls (R) • Message Functional Definition and Content (R) • General Message Format and Information Elements Coding (R) • Supplementary Services (C) • PCM-24 Digital Trunk Interface (R) • Interface Characteristics (R) • Supervisory Channel Associated Signaling (R) • Clear Channel Capability (R) • Alarm and Restoral Requirements (R) • PCM-30 Digital Trunk Interface (Europe only) (R) • Interoperation of PCM-24 and PCM-30 (R) • Analog Trunk Interface (C) • Integrated Digital Loop Carrier (C) • Trunk Group-Remove from Service (R) • Trunk Group-Restore to Service (R)	• UCR Section 5.2.1.3.1 • UCR Section 5.2.1.3.2 • UCR Section 5.2.1.3.4.1 • UCR Section 5.2.1.3.4.1.1 • UCR Section 5.2.1.3.4.2 • UCR Section 5.2.1.3.4.2.1 • UCR Section 5.2.4.3.3.1.1 • UCR Section 5.2.4.3.3.1.2 • UCR Section 5.2.4.3.3.2.1 • UCR Section 5..2.4.3.3.2.2 • UCR Section 5.2.4.3.5 • UCR Section 5.2.4.3.6 • UCR Section 5.2.3.4.7 • UCR Section 5.2.3.4.8 • UCR Section 5..2.3.4.9 • UCR Section 5.2.3.4.10 • UCR Section 5..2.4.4.1 • UCR Section 5.2.4.4.2 • UCR Section 5..2.4.4.2.1 • UCR Section 5.2.4.4.3 • UCR Section 5..2.4.5.1 • UCR Section 5.2.4.7.1.4.2 • UCR Section 5..2.4.7.1.1 • UCR Section 5.2.4.7.1.2 • UCR Section 5.2.4.7.1.3 • UCR Section 5..2.4.7.1.3.1 • UCR Section 5.2.4.7.1.3.2 • UCR Section 5.2.4.7.1.4 • UCR Section 5.2.4.7.1.4.2 • UCR Section 5.2.4.7.1.4.3 • UCR Section 5.2.4.7.1.4.4 • UCR Section 5.2.4.7.1.4.5 • UCR Section 5.2.4.7.1.4.6 • UCR Section 5.2.6.1 • UCR Section 5.2.6.1.1 • UCR Section 5.2.6.1.2 • UCR Section 5.2.6.1.3 • UCR Section 5.2.6.1.4 • UCR Section 5.2.6.2 • UCR Section 5.2.6.3 • UCR Section 5.2.6.4 • UCR Section 5.2.6.5 • UCR Section 5.2.1.5.5 • UCR Section 5.2.1.5.5

Table 3. PBX 1 Requirements (continued)

DSN Trunk Interfaces (continued)				
Interface	**Critical**	**Requirements Required or Conditional**		**References**
T1 CAS (MFR1, DTMF, DP) E1 CAS (MFR1, DTMF, DP) T1 ISDN PRI NI 1/2 (ANSI T1.619a) E1 ISDN PRI (ITU-T Q.955.3)	No No (Europe only) Yes No (Europe only)	Voice	• MOS (R) • Secure calls (R)	• CJCSI 6215.01C • CJCSI 6215.01C
		Facsimile	• Analog: ITU-T T.4 (R)	• DISR
		Data	• Modem (VBD) (R) • 56 kbps switched data (R: PRI only) • 64 kbps switched data (R: PRI only) • NX56 synchronous BER (R: PRI only) • NX64 synchronous BER (R: PRI only) • Secure data (STE/STU-III) (R)	• CJCSI 6215.01C • UCR Section 5.2.2.9.6 • UCR Section 5.2.2.9.6 • UCR Section 5.2.2.9.6 • UCR Section 5.2.2.9.6 • CJCSI 6215.01C
		VTC	• ITU-T H.320 (R: PRI only)	• FTR 1080B-2002
DSN Line Interfaces				
2-Wire Analog ISDN BRI NI 1/2 (ANSI T1.619a) 2-Wire Proprietary Digital VoIP (Ethernet IEEE 802.3u)	Yes No No No	Access	• Directory Number Identification (R) • Analog Line (R) • National ISDN 1/2 Basic Access (R: BRI Only) • Basic Line Test Capabilities (R) • Advanced Line Test Capabilities (C) • Loop Start Line (R: 2-Wire Analog only) • Reverse Battery (R: 2-WireAnalog only) • Alerting Signals and Tones (R) • S/T Reference Point (R: ISDN BRI only) • VoIP System Requirements (R: VoIP Phones only)	• UCR Section 5.2.1.1.1 • UCR Section 5.2.1.3.5 • UCR Section 5.2.1.3.3 • UCR Section 5.2.1.5.4.1.1 • UCR Section 5.2.1.5.4.1.1 • UCR Section 5.2.4.2.1 • UCR Section 5.2.4.3.1 • UCR Section 5.2.4.5.1 • UCR Section 5.2.4.7.1.2.1 • UCR Section 5.2.12.8
		Voice	• MOS (R) • Secure Calls (R)	• CJCSI 6215.01C • CJCSI 6215.01C
		Facsimile	• Analog: ITU-T T.4 (R)	• DISR
		Data	• Modem (VBD) (R: 2-Wire Analog only) • Secure data (STE/STU-III) (R: 2-Wire Analog only)	• CJCSI 6215.01C • CJCSI 6215.01C
		VTC	• ITU-T H.320 (R: BRI only)	• FTR 1080B-2002
DSN Features & Capabilities				
Feature/ Capability	**Critical**	**Requirements Required or Conditional**		**References**
Common Features	Yes	• Individual Lines (R) • Denied originating service (C) • Code restriction and diversion (R) • Call waiting (R) • Three-way calling (R) • Add-on transfer, conference calling, and call hold (C) • Call Transfer Individual – All calls (R) • Call Transfer - Internal Only (R) • Call Transfer – Individual – Incoming Only/Add-On Consultation Hold – Incoming Call (R) • Call Transfer – Outside (R) • Call Transfer – Add-On Restricted Station (C) • Call Transfer – Attendant (C) • Call Hold (R) • Conference Calling – Six Way Station Controlled (C) • Call Forwarding Variable (R) • Call Forward Busy Line (R) • Call Forwarding – Don’t Answer – All Calls (R) • Selective Call Forwarding (C) • Call pick-up (C) • Address Translation (C) • Assured Dial Tone (R)		• UCR Section 5.2.1.1.1 • UCR Section 5.2.1.1.3 • UCR Section 5.2.1.1.4 • UCR Section 5.2.1.1.5.1 • UCR Section 5.2.1.1.6 • UCR Section 5.2.1.1.7 • UCR Section 5.2.1.1.7.1 • UCR Section 5.2.1.1.7.2 • UCR Section 5.2.1.1.7.3 • UCR Section 5.2.1.1.7.4 • UCR Section 5.2.1.1.7.5 • UCR Section 5.2.1.1.7.6 • UCR Section 5.2.1.1.7.7 • UCR Section 5.2.1.1.7.8 • UCR Section 5.2.1.1.8.1 • UCR Section 5.2.1.1.8.2 • UCR Section 5.2.1.1.8.3 • UCR Section 5.2.1.1.8.4 • UCR Section 5.2.1.1.9.1 • UCR Section 5.2.1.7 • UCR Section 5.2.1.9
Attendant	No	• Attendant Features (C)		• UCR Section 5.2.1.2.2

Table 3. PBX 1 Requirements (continued)

DSN Features & Capabilities			
Feature/ Capability	**Critical**	**Requirements Required or Conditional**	**References**
Public Safety	Yes	• Emergency Service (911) Caller (R) • Emergency Service (911) Public Safety Answering Service (C) • Enhanced Emergency Service (E911) (C) • Trace of terminating calls (R) • Outgoing call trace (R)	• UCR Section 5.2.1.4.1.1 • UCR Section 5.2.1.4.1.2 • UCR Section 5.2.1.4.1.3 • UCR Section 5.2.1.4.2 • UCR Section 5.2.1.4.3
Conferencing	No	• Preset Conferencing (C) • Meet-Me Conferencing (C) • Progressive Conferencing (C)	• UCR Section 5.2.1.6.1 • UCR Section 5.2.1.6.2 • UCR Section 5.2.1.6.3
Nailed-up Connections	No	• Nailed-Up Connections (C)	• UCR Section 5.2.1.8
DSN Hotline Services	No	• DSN Analog Hotline Service (C)	• UCR Section 5.2.1.12
MLPP	Yes	• MLPP Overview (R) • Preemption in the Network (R) • Network Facility with Lower Precedence Calls (R) • Network Facility with Equal or Higher Precedence Calls (R) • Precedence Call Diversion (R) • Channel Associated Signaling (R) • Primary Rate Interface (R) • Analog Line MLPP (R) • ISDN MLPP Basic Rate Interface (R) • ISDN Primary Rate Interface (R) • Precedence Call Waiting (R) • Call Forwarding (R) • Call Transfer (R) • Call Hold (R) • Three-Way Calling (R) • Call Pickup (C) • Conferencing (C) • Multiline Hunt Group (C) • Community of Interest (C) • MLPP Interaction with EKTS features (C)	• UCR Section 5.2.2.1.1 • UCR Section 5.2.2.2 • UCR Section 5.2.2.2.1 • UCR Section 5.2.2.2.2 • UCR Section 5.2.2.3 • UCR Section 5.2.2.4.1 • UCR Section 5.2.2.4.2 • UCR Section 5.2.2.5 • UCR Section 5.2.2.6 • UCR Section 5.2.2.7 • UCR Section 5.2.2.8.1 • UCR Section 5.2.2.8.2 • UCR Section 5.2.2.8.3 • UCR Section 5.2.2.8.4 • UCR Section 5.2.2.8.5 • UCR Section 5.2.2.8.6 • UCR Section 5.2.2.8.7.1 • UCR Section 5.2.2.8.8 • UCR Section 5.2.2.8.9 • UCR Section 5.2.2.10.1

Table 3. PBX 1 Requirements (continued)

DSN Features & Capabilities (continued)			
Feature/ Capability	**Critical**	**Requirements Required or Conditional**	**References**
Call Processing	Yes	• Call Treatments (R) • Primary and Alternate Routing (R) • E&M Lead Signaling States (C) • 4-Wire Analog User Access Lines (C) • 2-Wire User Access Lines (R) • Termination of Analog Lines (R) • DSN User Dialing (R) • Interswitch and Intraswitch Dialing (R) • Seven-Digit Dialing (R) • Ten-Digit Dialing (R) • Access Code (R) • Access Digit (R) • Precedence Digit (R) • Service Digit (R) • Route Code (R) • Area Code (R) • Switch Code (R) • Line Number (R) • Calling Name Delivery (C) • Calling Number Delivery (R) • Emergency Service 911 Conflict Resolution (R) • DSN Switch Outpulsing Digit Formats (C) • Standard Directory Number (R) • Standard Test Numbers (C) • Base Services – Abbreviated Numbers (R) • Digit Reception Requirements (R) • Screening (R)	• UCR Section 5.2.3.1 • UCR Section 5.2.3.2 • UCR Section 5.2.3.3.1 • UCR Section 5.2.3.3.2 • UCR Section 5.2.3.3.3 • UCR Section 5.2.3.3.4 • UCR Section 5.2.3.5.1.1 • UCR Section 5.2.3.5.1.1 • UCR Section 5.3.3.5.2.1 • UCR Section 5.2.3.5.2.2 • UCR Section 5.2.3.5.1.3 • UCR Section 5.2.3.5.1.3.1 • UCR Section 5.2.3.5.1.3.2 • UCR Section 5.2.3.5.1.3.3 • UCR Section 5.2.3.5.1..4 • UCR Section 5.2.3.5.1.5 • UCR Section 5.2.3.5.1.6 • UCR Section 5.2.3.5.1.7 • UCR Section 5.2.3.5.1.8.1 • UCR Section 5.2.3.5.1.8.2 • UCR Section 5.2.3.5.1.9 • UCR Section 5.2.3.5.2 • UCR Section 5.2.3.5.3 • UCR Section 5.2.3.5.4 • UCR Section 5.2.3.5.5 • UCR Section 5.2.3.5.6 • UCR Section 5.2.3.5.8
ISDN Services	Yes	• BRI Access, Call Control and Signaling (R) • Uniform Interface Configuration for BRIs (R) • EKTS (C) • PRI Access, Call Control and Signaling (R) • PRI Features (R) • Packet Data Features and Capabilities (C)	• UCR Section 5.2.9.2, Table 5.2.9-1 • UCR Section 5.2.9.2, Table 5.2.9-2 • UCR Section 5.2.9.3, Table 5.2.9-3 • UCR Section 5.2.9.2, Table 5.2.9-4 • UCR Section 5.2.9.2, Table 5.2.9-5 • UCR Section 5.2.9.2, Table 5.2.9-6
Synchronization	Yes	• Line timing mode (R) • Internal Stratum 4 (R) • Synchronization Performance Monitoring Criteria (C) • DS1 Traffic Interfaces (C) • DS0 Traffic Interconnects (C)	• UCR Section 5.2.10.1.1.2 • UCR Section 5.2.10.1.1.2.2 • UCR Section 5.2.10.2 • UCR Section 5.2.10.3 • UCR Section 5.2.10.4
Reliability	Yes	• System Availability (R) • Backup Power (R) • Power Components (R) • UPS Requirements (R) • UPS PBX 1 Load Capacity (R) • Backup Power (Environmental) (R) • Alarms (R)	• UCR Section 5.2.11.2 • UCR Section 5.2.11.3 • UCR Section 5.2.11.3.1 • UCR Section 5.2.11.3.2 • UCR Section 5.2.11.3.2.1 • UCR Section 5.2.11.3.3 • UCR Section 5.2.11.3.4
Security	Yes	• GR-815, STIGs, and DoDI 8510.bb (DIACAP) (R)	• UCR Section 3

Table 3. PBX 1 Requirements (continued)

VoIP			
Feature/ Capability	**Critical**	**Requirements Required or Conditional**	**References**
VoIP System	No	VoIP function is conditional. If VoIP is provided, **all** of the following requirements must be met: • Voice Quality with MOS of 4.0 or better (R) • ITU-T G.711 PCM CODEC (R) • MLPP (R) • Security (R) • Network management (C) • System timing (R) • Latency ≤ 60 milliseconds (R) • IPv6 capable (R) • Service Class Tagging (R) • VoIP System Downtime (IP network 80 min/yr Subscriber 20 min/yr) (R) • Softphone Requirements	• UCR section 5.2.12.8.2.1 • UCR section 5.2.12.8.2.2 • UCR section 5.2.12.8.2.3 • UCR section 5.2.12.8.2.4 • UCR section 5.2.12.8.2.5 • UCR section 5.2.12.8.2.6 • UCR section 5.2.12.8.2.7 • UCR section 5.2.12.8.2.8 • UCR section 5.2.12.8.2.9 • UCR section 5.2.12.8.2.10 • DISA Memo Reference (h)

Network Gateways				
Gateway	**Critical**	**Requirements Required or Conditional**		**References**
PSTN (See note.)	No	Trunking	• Positive Identification Control (C) • On-Netting (C) • Off-Netting (C) • Ground Start Line (R) • Immediate Start (C) • Delay Dial (C)	• CJCSI 6215.01C • CJCSI 6215.01C • CJCSI 6215.01C • UCR Section 5.2.4.2.2 • UCR Section 5.2.4.3.2 • UCR Section 5.2.4.3.4

NOTE: Voice, facsimile, data, and VTC service requirements for PSTN are identical to DSN with the exception of MLPP.

Table 3. PBX 1 Requirements (continued)

LEGEND:

802.3u	Standard for carrier sense multiple access with collision detection at 100 Mbps	FTR 1080B-2002	Video Teleconferencing Services	PCM-24	Pulse Code Modulation - 24 Channels
ANSI	American National Standards Institute	G.711	PCM of voice frequencies	PCM-30	Pulse Code Modulation - 30 Channels
BER	Bit Error Ratio	GR	Generic Requirement	PRI	Primary Rate Interface
BRI	Basic Rate Interface	GR-815	Generic Requirements For Network Element/Network System (NE/NS) Security	PSTN	Public Switched Telephone Network
C	Conditional	H.320	Standard for Narrowband VTC	Q.955.3	ISDN Signaling Standard for E1 MLPP
CAS	Channel Associated Signaling	IEEE	Institute of Electrical and Electronics Engineers	R	Required
CJCSI	Chairman of the Joint Chiefs of Staff Instruction	IP	Internet Protocol	S/T	ISDN BRI four-wire interface
CODEC	Coder/Decoder	IPv6	Internet Protocol version 6	SS7	Signaling System 7
DIACAP	DoD Information Assurance Certification and Accreditation Process	ISDN	Integrated Services Digital Network	STE	Secure Terminal Equipment
DISA	Defense Information Systems Agency	IT	Information Technology	STIGs	Security Technical Implementation Guides
DISR	DoD IT Standards Registry	ITU-T	International Telecommunication Union-Telecommunication Standardization Sector	STU-III	Secure Telephone Unit -3rd generation
DoD	Department of Defense	kbps	kilobits per second	T.4	Standardization of Group 3 facsimile terminals for document transmission
DoDI	Department of Defense Instruction	Mbps	Megabits per second	T1	Digital Transmission Link Level 1 (1.544 Mbps)
DP	Dial Pulse	MFR1	Multi-Frequency Recommendation 1	T1.619a	SS7 and ISDN MLPP Signaling Standard for T1
DS0	Digital Signal Level 0 (64 kbps)	min	minute	UCR	Unified Capabilities Requirements
DS1	Digital Signal Level 1 (1.544 Mbps) (2.048 Mbps European)	MLPP	Multi-Level Precedence and Preemption	UPS	Uninterruptible Power Supply
DSN	Defense Switched Network	MOS	Mean Opinion Score	VBD	Variable bit data
DTMF	Dual Tone Multi-Frequency	NI 1/2	National ISDN Standard 1 or 2	VoIP	Voice over Internet Protocol
E&M	Ear and Mouth	NX56	Data format restricted to multiples of 56 kbps	VTC	Video Teleconferencing
E1	European Basic Multiplex Rate (2.048 Mbps)	NX64	Data format restricted to multiples of 64 kbps	yr	year
EKTS	Electronic Key Telephone System	PBX	Private Branch Exchange		
FTR	Federal Telecommunications Recommendation	PBX 1	Private Branch Exchange 1		
		PCM	Pulse Code Modulation		

5. No detailed test report was developed in accordance with the Program Manager's request. JITC distributes interoperability information via the JITC Electronic Report Distribution (ERD) system, which uses Unclassified-But-Sensitive Internet Protocol Router Network (NIPRNet) e-mail. More comprehensive interoperability status information is available via the JITC System Tracking Program (STP). The STP is accessible by .mil/gov users on the NIPRNet at https://stp.fhu.disa.mil. Test reports, lessons learned, and related testing documents and references are on the JITC Joint Interoperability Tool (JIT) at http://jit.fhu.disa.mil (NIPRNet), or http://199.208.204.125 (SIPRNet). Information related to DSN testing is on the Telecom Switched Services Interoperability (TSSI) website at http://jitc.fhu.disa.mil/tssi. Due to the sensitivity of the information, the Information Assurance Accreditation Package (IAAP) that contains the approved configuration and deployment guide must be requested directly through government civilian or uniformed military personnel from the Unified Capabilities Certification Office (UCCO), e-mail: ucco@disa.mil.

6. The JITC point of contact is Mr. Edward Mellon, DSN 879-5159, commercial (520) 538-5159, FAX DSN 879-4347, or e-mail to edward.mellon@disa.mil. The JITC's mailing address is P.O. Box 12798, Fort Huachuca, AZ 85670-2798. The tracking number for the SUT is 0901201. The tracking number for the Cisco Internet Protocol Communicator is 0911101.

FOR THE COMMANDER:

J. T. Schutte

2 Enclosures a/s

for RICHARD A. MEADOR
Chief
Battlespace Communications Portfolio

Distribution (electronic mail):
Joint Staff J-6
Joint Interoperability Test Command, Liaison, TE3/JT1
Office of Chief of Naval Operations, CNO N6F2
Headquarters U.S. Air Force, Office of Warfighting Integration & CIO, AF/XCIN (A6N)
Department of the Army, Office of the Secretary of the Army, DA-OSA CIO/G-6 ASA (ALT), SAIS-IOQ
U.S. Marine Corps MARCORSYSCOM, SIAT, MJI Division I
DOT&E, Net-Centric Systems and Naval Warfare
U.S. Coast Guard, CG-64
Defense Intelligence Agency
National Security Agency, DT
Defense Information Systems Agency, TEMC
Office of Assistant Secretary of Defense (NII)/DOD CIO
U.S. Joint Forces Command, Net-Centric Integration, Communication, and Capabilities Division, J68
Defense Information Systems Agency, GS23

ADDITIONAL REFERENCES

(c) Joint Interoperability Test Command, "Information Assurance (IA) Assessment of Cisco Unified Communication Manager Version 7.1(2) with Internetwork Operating System (IOS) Software Release 12.4(22)T2 (TN0901201)," 10 March 2010
(d) Chairman of the Joint Chiefs of Staff Instruction (CJCSI) 6215.01C, "Policy for Department of Defense Voice Services with Real Time Services (RTS)," 9 November 2007
(e) Defense Information Systems Agency, "Department of Defense Networks Unified Capabilities Requirements," December 2008
(f) Joint Interoperability Test Command, "Defense Switched Network Generic Switch Test Plan (GSTP), Change 2," 2 October 2006
(g) Office of the Secretary of Defense, "Interim Unified Capabilities (UC) IPv6 Rules of Engagement (ROE)," 31 July 2009
(h) Defense Information Systems Agency NS3 Memorandum, "Softphone Certification" 20 April 2009

CERTIFICATION TESTING SUMMARY

1. SYSTEM TITLE. Unified Communications Manager Version 7.1(2), with Internetwork Operating System (IOS) Software Release 12.4(22) T2; hereinafter referred to as the System Under Test (SUT).

2. PROPONENT. Headquarters United States Air Force Europe (HQ USAFE).

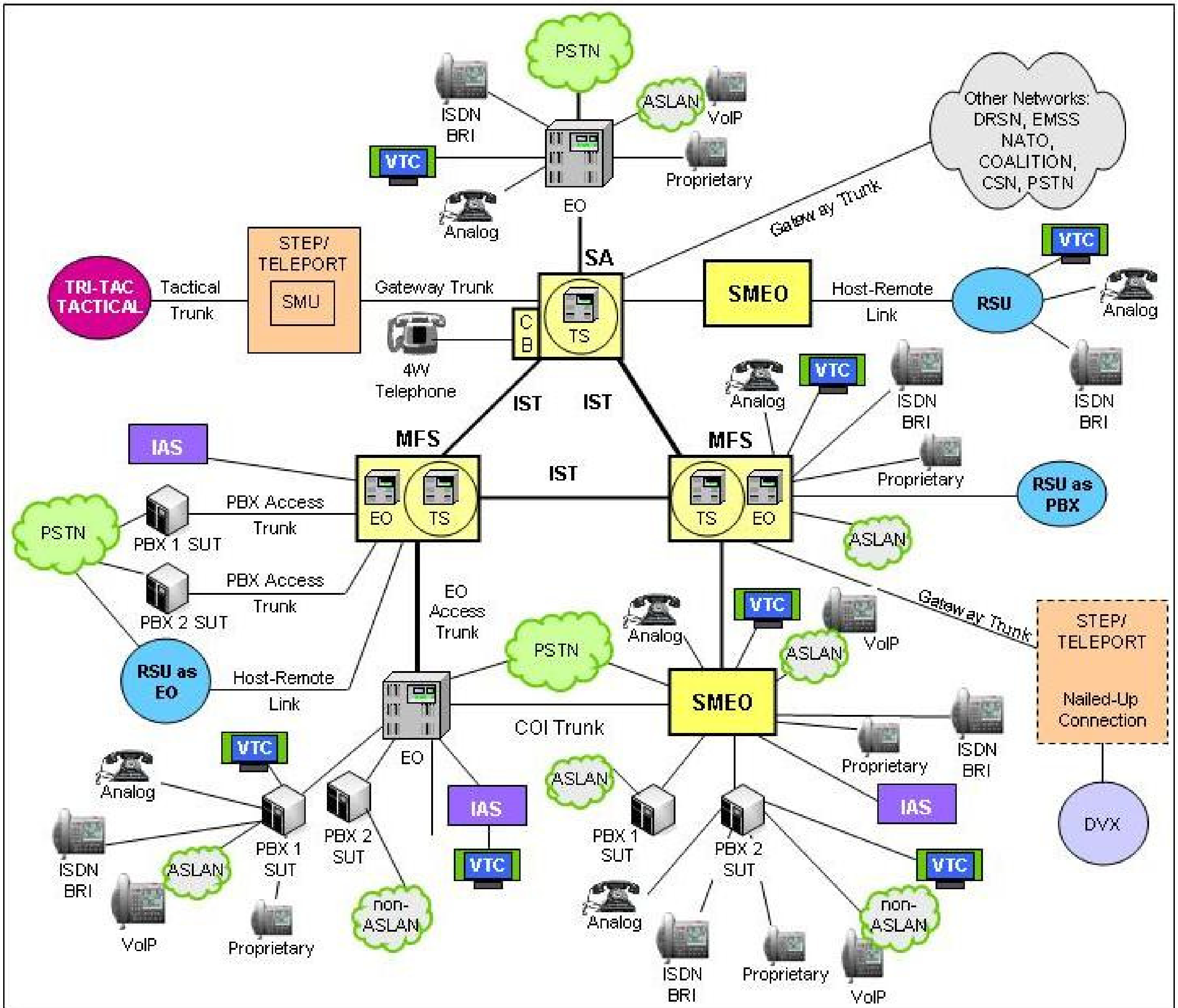
3. PROGRAM MANAGER. Joseph Halcli, HQ USAFE/A6NA, PSC2 Box 11095, APO AE, 09012, e-mail: joseph.halcli@ramstein.af.mil.

4. TESTER. Joint Interoperability Test Command (JITC), Fort Huachuca, Arizona.

5. SYSTEM UNDER TEST DESCRIPTION. The SUT is a Private Branch Exchange (PBX) 1. The SUT supports American National Standards Institute (ANSI) T1.619a Digital Transmission Link Level 1 (T1) Integrated Services Digital Network (ISDN) Primary Rate Interface (PRI) National ISDN Standard 1 or 2 (NI 1/2) and International Telecommunication Union - Telecommunication Standardization Sector (ITU-T) Q.931 European Basic Multiplex Rate (E1) ISDN PRI interfaces. The SUT consists of Communication Managers running the Cisco Unified Communication Manager software, gateways, and Internet Protocol (IP) telephones. The Cisco Unified Communication Manager is the software-based call-processing component of the Cisco enterprise IP telephone solution. The Cisco Unified Communication Manager software is a client-server application loaded on Cisco 7800 Series Media Convergence Servers (MCSs). The Cisco Communication Manager software provides telephony features and capabilities to packet telephony network devices such as VoIP phones. The Cisco Unified Communication Managers tested were the MCS7835I2, MCS7835H2, MCS7825H3, and MCS7825H4. The other family series of servers which include: the MCS7835H1, MCS7845H1, MCS7845H2, MCS7825I4, MCS7835I1, MCS7845I1, and MCS7845I2 utilize the same software and similar hardware and JITC analysis determined them to be functionally identical for interoperability certification purposes and they are also certified for joint use.

The 2851 and 3845 scalable integrated services routers are included in this tested architecture. The 2851 has one Network Module (NM) slot, one High-Density Extension Voice Module (EVM-HD) slot, and four High-Performance Wide Area Network (WAN) Interface Card (WIC) (HWIC) slots. These slots can be populated with up to 12 T1 trunks or 52 Foreign Exchange Station (FXS) ports. The 2811 and 2821 utilize the same software and similar hardware as the 2851 and JITC analysis determined them to be functionally identical for interoperability certification purposes and they are also certified for joint use. The 3845 has four NM slots and four HWIC slots. Each NM slot on the 3845 can accommodate a standard NM, an enhanced-network-module (NME) or an EVM-HD. The 3845 supports up to 24 T1 trunks or 88 FXS ports. The 3825 utilizes the same software and similar hardware as the 3845 and JITC analysis determined it to be functionally identical for interoperability certification purposes and it is also certified for joint use.

6. OPERATIONAL ARCHITECTURE. The Defense Switched Network (DSN) architecture is a two-level network hierarchy consisting of DSN backbone switches and Service/Agency installation switches. Joint Staff policy and subscriber mission requirements determine which type of switch can be used at a particular location. The DSN architecture, therefore, consists of several categories of switches including PBXs. The Unified Capabilities Requirements (UCR) operational DSN Architecture is depicted in Figure 2-1. The architecture depicts the relationship of Military Department PBX 1s to the other DSN switch types.



LEGEND:

4W	4-Wire
ASLAN	Assured Services Local Area Network
BRI	Basic Rate Interface
CB	Channel Bank
COI	Community of Interest
CSN	Canadian Switch Network
DRSN	Defense Red Switch Network
DSN	Defense Switched Network
DVX	Deployable Voice Exchange
EMSS	Enhanced Mobile Satellite System
EO	End Office
IAS	Integrated Access Switch
ISDN	Integrated Services Digital Network
IST	Interswitch Trunk
MFS	Multifunction Switch
NATO	North Atlantic Treaty Organization
PBX	Private Branch Exchange
PBX 1	Private Branch Exchange 1
PBX 2	Private Branch Exchange 2
PSTN	Public Switched Telephone Network
RSU	Remote Switching Unit
SA	Standalone
SMEO	Small End Office
SMU	Switched Multiplex Unit
STEP	Standardized Tactical Entry Point
SUT	System Under Test
Tri-Tac	Tri-Service Tactical Communications Program
TS	Tandem Switch
VoIP	Voice over Internet Protocol
VTC	Video Teleconferencing

Figure 2-1. DSN Architecture

7. REQUIRED SYSTEM INTERFACES. Requirements specific to PBX 1s are listed in Table 2-1. These requirements are derived from:

a. DSN services for Network and Applications specified in Chairman of the Joint Chiefs of Staff Instruction (CJCSI) 6215.01C, “Policy for Department of Defense Voice Services with Real Time Services (RTS)”, Reference (d).

b. UCR interface and signaling requirements for trunks/lines verified through JITC testing and/or vendor submission of Letters of Compliance (LoC), Reference (e).

c. UCR PBX 1 Capability Requirements (CRs) and Feature Requirements (FRs) verified through JITC testing and/or vendor submission of LoC, Reference (e).

d. The IPv6 requirements specified in References (e) and (g).

e. The softphone requirements specified in Reference (h).

Table 2-1. PBX 1 Requirements

DSN Trunk Interfaces				
Interface	**Critical**	**Requirements Required or Conditional**		**References**
			• PBX Line (C)	• UCR Section 5.2.1.3.1
			• Direct Inward Dialing (C)	• UCR Section 5.2.1.3.2
			• National ISDN 1/2 Primary Access (R)	• UCR Section 5.2.1.3.4.1
			• ISDN ANSI MLPP Service Capability (R)	• UCR Section 5.2.1.3.4.1.1
			• ITU-T ISDN Primary Access (Europe only) (C)	• UCR Section 5.2.1.3.4.2
			• ITU-T ISDN Primary Access Digital Subscriber Signaling System Number 1 MLPP (Europe only) (C)	• UCR Section 5.2.1.3.4.2.1
			• Normal Wink Start Operations (R)	• UCR Section 5.2.4.3.3.1.1
T1 CAS (MFR1, DTMF, DP)	No		• Glare Operation (R)	• UCR Section 5.2.4.3.3.1.2
			• Abnormal Wink Start (R)	• UCR Section 5.2.4.3.3.2.1
			• Glare Resolution (R)	• UCR Section 5..2.4.3.3.2.2
			• Call for Service Timing (R)	• UCR Section 5.2.4.3.5
			• Guard Timing (R)	• UCR Section 5.2.4.3.6
			• Satellite Timing (R)	• UCR Section 5.2.3.4.7
			• Disconnect Control (R)	• UCR Section 5.2.3.4.8
			• Reselect and Retrial (R)	• UCR Section 5..2.3.4.9
			• Off-Hook Supervision Transition (R)	• UCR Section 5.2.3.4.10
E1 CAS (MFR1, DTMF, DP)	No (Europe only)		• Dial-Pulse Signals (R)	• UCR Section 5..2.4.4.1
			• DTMF Signaling (R)	• UCR Section 5.2.4.4.2
			• Standard Digit Format for Precedence (C)	• UCR Section 5..2.4.4.2.1
			• MFR1 2/6 Signaling (C)	• UCR Section 5.2.4.4.3
			• Alerting Signals and Tones (R)	• UCR Section 5..2.4.5.1
			• DSN ISDN User-to-Network Signaling (R)	• UCR Section 5.2.4.7.1.4.2
		Trunking	• Application (R)	• UCR Section 5..2.4.7.1.1
			• Physical Layer (R)	• UCR Section 5.2.4.7.1.2
			• Data Link Layer (R)	• UCR Section 5.2.4.7.1.3
T1 ISDN PRI NI 1/2 (ANSI T1.619a)	Yes		• Data Link Connection (R)	• UCR Section 5..2.4.7.1.3.1
			• Peer-to-Peer Procedures of Data-Link Layer (R)	• UCR Section 5.2.4.7.1.3.2
			• Layer 3 DSN User-to-Network Signaling (R)	• UCR Section 5.2.4.7.1.4
			• DSN User-to-Network Signaling for Circuit-Switched Bearer Services (R)	• UCR Section 5.2.4.7.1.4.2
			• Sequence of Messages for DSN Circuit-Switched Calls (R)	• UCR Section 5.2.4.7.1.4.3
			• Message Functional Definition and Content (R)	• UCR Section 5.2.4.7.1.4.4
			• General Message Format and Information Elements Coding (R)	• UCR Section 5.2.4.7.1.4.5
E1 ISDN PRI (ITU-T Q.955.3)	No (Europe only)		• Supplementary Services (C)	• UCR Section 5.2.4.7.1.4.6
			• PCM-24 Digital Trunk Interface (R)	• UCR Section 5.2.6.1
			• Interface Characteristics (R)	• UCR Section 5.2.6.1.1
			• Supervisory Channel Associated Signaling (R)	• UCR Section 5.2.6.1.2
			• Clear Channel Capability (R)	• UCR Section 5.2.6.1.3
			• Alarm and Restoral Requirements (R)	• UCR Section 5.2.6.1.4
			• PCM-30 Digital Trunk Interface (Europe only) (R)	• UCR Section 5.2.6.2
			• Interoperation of PCM-24 and PCM-30 (R)	• UCR Section 5.2.6.3
			• Analog Trunk Interface (C)	• UCR Section 5.2.6.4
			• Integrated Digital Loop Carrier (C)	• UCR Section 5.2.6.5
			• Trunk Group-Remove from Service (R)	• UCR Section 5.2.1.5.5
			• Trunk Group-Restore to Service (R)	• UCR Section 5.2.1.5.5

Table 2-1. PBX 1 Requirements (continued)

DSN Trunk Interfaces (continued)				
Interface	**Critical**	**Requirements Required or Conditional**		**References**
T1 CAS (MFR1, DTMF, DP) E1 CAS (MFR1, DTMF, DP) T1 ISDN PRI NI 1/2 (ANSI T1.619a) E1 ISDN PRI (ITU-T Q.955.3)	No No (Europe only) Yes No (Europe only)	Voice	• MOS (R) • Secure calls (R)	• CJCSI 6215.01C • CJCSI 6215.01C
		Facsimile	• Analog: ITU-T T.4 (R)	• DISR
		Data	• Modem (VBD) (R) • 56 kbps switched data (R: PRI only) • 64 kbps switched data (R: PRI only) • NX56 synchronous BER (R: PRI only) • NX64 synchronous BER (R: PRI only) • Secure data (STE/STU-III) (R)	• CJCSI 6215.01C • UCR Section 5.2.2.9.6 • UCR Section 5.2.2.9.6 • UCR Section 5.2.2.9.6 • UCR Section 5.2.2.9.6 • CJCSI 6215.01C
		VTC	• ITU-T H.320 (R: PRI only)	• FTR 1080B-2002
DSN Line Interfaces				
2-Wire Analog ISDN BRI NI 1/2 (ANSI T1.619a) 2-Wire Proprietary Digital VoIP (Ethernet IEEE 802.3u)	Yes No No No	Access	• Directory Number Identification (R) • Analog Line (R) • National ISDN 1/2 Basic Access (R: BRI Only) • Basic Line Test Capabilities (R) • Advanced Line Test Capabilities (C) • Loop Start Line (R: 2-Wire Analog only) • Reverse Battery (R: 2-WireAnalog only) • Alerting Signals and Tones (R) • S/T Reference Point (R: ISDN BRI only) • VoIP System Requirements (R: VoIP Phones only)	• UCR Section 5.2.1.1.1 • UCR Section 5.2.1.3.5 • UCR Section 5.2.1.3.3 • UCR Section 5.2.1.5.4.1.1 • UCR Section 5.2.1.5.4.1.1 • UCR Section 5.2.4.2.1 • UCR Section 5.2.4.3.1 • UCR Section 5.2.4.5.1 • UCR Section 5.2.4.7.1.2.1 • UCR Section 5.2.12.8
		Voice	• MOS (R) • Secure Calls (R)	• CJCSI 6215.01C • CJCSI 6215.01C
		Facsimile	• Analog: ITU-T T.4 (R)	• DISR
		Data	• Modem (VBD) (R: 2-Wire Analog only) • Secure data (STE/STU-III) (R: 2-Wire Analog only)	• CJCSI 6215.01C • CJCSI 6215.01C
		VTC	• ITU-T H.320 (R: BRI only)	• FTR 1080B-2002
DSN Features & Capabilities				
Feature/ Capability	**Critical**	**Requirements Required or Conditional**		**References**
Common Features	Yes	• Individual Lines (R) • Denied originating service (C) • Code restriction and diversion (R) • Call waiting (R) • Three-way calling (R) • Add-on transfer, conference calling, and call hold (C) • Call Transfer Individual – All calls (R) • Call Transfer - Internal Only (R) • Call Transfer – Individual – Incoming Only/Add-On Consultation Hold – Incoming Call (R) • Call Transfer – Outside (R) • Call Transfer – Add-On Restricted Station (C) • Call Transfer – Attendant (C) • Call Hold (R) • Conference Calling – Six Way Station Controlled (C) • Call Forwarding Variable (R) • Call Forward Busy Line (R) • Call Forwarding – Don't Answer – All Calls (R) • Selective Call Forwarding (C) • Call pick-up (C) • Address Translation (C) • Assured Dial Tone (R)		• UCR Section 5.2.1.1.1 • UCR Section 5.2.1.1.3 • UCR Section 5.2.1.1.4 • UCR Section 5.2.1.1.5.1 • UCR Section 5.2.1.1.6 • UCR Section 5.2.1.1.7 • UCR Section 5.2.1.1.7.1 • UCR Section 5.2.1.1.7.2 • UCR Section 5.2.1.1.7.3 • UCR Section 5.2.1.1.7.4 • UCR Section 5.2.1.1.7.5 • UCR Section 5.2.1.1.7.6 • UCR Section 5.2.1.1.7.7 • UCR Section 5.2.1.1.7.8 • UCR Section 5.2.1.1.8.1 • UCR Section 5.2.1.1.8.2 • UCR Section 5.2.1.1.8.3 • UCR Section 5.2.1.1.8.4 • UCR Section 5.2.1.1.9.1 • UCR Section 5.2.1.7 • UCR Section 5.2.1.9
Attendant	No	• Attendant Features (C)		• UCR Section 5.2.1.2.2

Table 2-1. PBX 1 Requirements (continued)

DSN Features & Capabilities			
Feature/ Capability	**Critical**	**Requirements Required or Conditional**	**References**
Public Safety	Yes	• Emergency Service (911) Caller (R) • Emergency Service (911) Public Safety Answering Service (C) • Enhanced Emergency Service (E911) (C) • Trace of terminating calls (R) • Outgoing call trace (R)	• UCR Section 5.2.1.4.1.1 • UCR Section 5.2.1.4.1.2 • UCR Section 5.2.1.4.1.3 • UCR Section 5.2.1.4.2 • UCR Section 5.2.1.4.3
Conferencing	No	• Preset Conferencing (C) • Meet-Me Conferencing (C) • Progressive Conferencing (C)	• UCR Section 5.2.1.6.1 • UCR Section 5.2.1.6.2 • UCR Section 5.2.1.6.3
Nailed-up Connections	No	• Nailed-Up Connections (C)	• UCR Section 5.2.1.8
DSN Hotline Services	No	• DSN Analog Hotline Service (C)	• UCR Section 5.2.1.12
MLPP	Yes	• MLPP Overview (R) • Preemption in the Network (R) • Network Facility with Lower Precedence Calls (R) • Network Facility with Equal or Higher Precedence Calls (R) • Precedence Call Diversion (R) • Channel Associated Signaling (R) • Primary Rate Interface (R) • Analog Line MLPP (R) • ISDN MLPP Basic Rate Interface (R) • ISDN Primary Rate Interface (R) • Precedence Call Waiting (R) • Call Forwarding (R) • Call Transfer (R) • Call Hold (R) • Three-Way Calling (R) • Call Pickup (C) • Conferencing (C) • Multiline Hunt Group (C) • Community of Interest (C) • MLPP Interaction with EKTS features (C)	• UCR Section 5.2.2.1.1 • UCR Section 5.2.2.2 • UCR Section 5.2.2.2.1 • UCR Section 5.2.2.2.2 • UCR Section 5.2.2.3 • UCR Section 5.2.2.4.1 • UCR Section 5.2.2.4.2 • UCR Section 5.2.2.5 • UCR Section 5.2.2.6 • UCR Section 5.2.2.7 • UCR Section 5.2.2.8.1 • UCR Section 5.2.2.8.2 • UCR Section 5.2.2.8.3 • UCR Section 5.2.2.8.4 • UCR Section 5.2.2.8.5 • UCR Section 5.2.2.8.6 • UCR Section 5.2.2.8.7.1 • UCR Section 5.2.2.8.8 • UCR Section 5.2.2.8.9 • UCR Section 5.2.2.10.1

Table 2-1. PBX 1 Requirements (continued)

DSN Features & Capabilities (continued)			
Feature/ Capability	**Critical**	**Requirements Required or Conditional**	**References**
Call Processing	Yes	• Call Treatments (R) • Primary and Alternate Routing (R) • E&M Lead Signaling States (C) • 4-Wire Analog User Access Lines (C) • 2-Wire User Access Lines (R) • Termination of Analog Lines (R) • DSN User Dialing (R) • Interswitch and Intraswitch Dialing (R) • Seven-Digit Dialing (R) • Ten-Digit Dialing (R) • Access Code (R) • Access Digit (R) • Precedence Digit (R) • Service Digit (R) • Route Code (R) • Area Code (R) • Switch Code (R) • Line Number (R) • Calling Name Delivery (C) • Calling Number Delivery (R) • Emergency Service 911 Conflict Resolution (R) • DSN Switch Outpulsing Digit Formats (C) • Standard Directory Number (R) • Standard Test Numbers (C) • Base Services – Abbreviated Numbers (R) • Digit Reception Requirements (R) • Screening (R)	• UCR Section 5.2.3.1 • UCR Section 5.2.3.2 • UCR Section 5.2.3.3.1 • UCR Section 5.2.3.3.2 • UCR Section 5.2.3.3.3 • UCR Section 5.2.3.3.4 • UCR Section 5.2.3.5.1.1 • UCR Section 5.2.3.5.1.1 • UCR Section 5.3.3.5.2.1 • UCR Section 5.2.3.5.2.2 • UCR Section 5.2.3.5.1.3 • UCR Section 5.2.3.5.1.3.1 • UCR Section 5.2.3.5.1.3.2 • UCR Section 5.2.3.5.1.3.3 • UCR Section 5.2.3.5.1..4 • UCR Section 5.2.3.5.1.5 • UCR Section 5.2.3.5.1.6 • UCR Section 5.2.3.5.1.7 • UCR Section 5.2.3.5.1.8.1 • UCR Section 5.2.3.5.1.8.2 • UCR Section 5.2.3.5.1.9 • UCR Section 5.2.3.5.2 • UCR Section 5.2.3.5.3 • UCR Section 5.2.3.5.4 • UCR Section 5.2.3.5.5 • UCR Section 5.2.3.5.6 • UCR Section 5.2.3.5.8
ISDN Services	Yes	• BRI Access, Call Control and Signaling (R) • Uniform Interface Configuration for BRIs (R) • EKTS (C) • PRI Access, Call Control and Signaling (R) • PRI Features (R) • Packet Data Features and Capabilities (C)	• UCR Section 5.2.9.2, Table 5.2.9-1 • UCR Section 5.2.9.2, Table 5.2.9-2 • UCR Section 5.2.9.3, Table 5.2.9-3 • UCR Section 5.2.9.2, Table 5.2.9-4 • UCR Section 5.2.9.2, Table 5.2.9-5 • UCR Section 5.2.9.2, Table 5.2.9-6
Synchronization	Yes	• Line timing mode (R) • Internal Stratum 4 (R) • Synchronization Performance Monitoring Criteria (C) • DS1 Traffic Interfaces (C) • DS0 Traffic Interconnects (C)	• UCR Section 5.2.10.1.1.2 • UCR Section 5.2.10.1.1.2.2 • UCR Section 5.2.10.2 • UCR Section 5.2.10.3 • UCR Section 5.2.10.4
Reliability	Yes	• System Availability (R) • Backup Power (R) • Power Components (R) • UPS Requirements (R) • UPS PBX 1 Load Capacity (R) • Backup Power (Environmental) (R) • Alarms (R)	• UCR Section 5.2.11.2 • UCR Section 5.2.11.3 • UCR Section 5.2.11.3.1 • UCR Section 5.2.11.3.2 • UCR Section 5.2.11.3.2.1 • UCR Section 5.2.11.3.3 • UCR Section 5.2.11.3.4
Security	Yes	• GR-815, STIGs, and DoDI 8510.bb (DIACAP) (R)	• UCR Section 3

Table 2-1. PBX 1 Requirements (continued)

VoIP			
Feature/ Capability	**Critical**	**Requirements Required or Conditional**	**References**
VoIP System	No	VoIP function is conditional. If VoIP is provided, **all** of the following requirements must be met: • Voice Quality with MOS of 4.0 or better (R) • ITU-T G.711 PCM CODEC (R) • MLPP (R) • Security (R) • Network management (C) • System timing (R) • Latency ≤ 60 milliseconds (R) • IPv6 capable (R) • Service Class Tagging (R) • VoIP System Downtime (IP network 80 min/yr Subscriber 20 min/yr) (R) • Softphone Requirements	• UCR section 5.2.12.8.2.1 • UCR section 5.2.12.8.2.2 • UCR section 5.2.12.8.2.3 • UCR section 5.2.12.8.2.4 • UCR section 5.2.12.8.2.5 • UCR section 5.2.12.8.2.6 • UCR section 5.2.12.8.2.7 • UCR section 5.2.12.8.2.8 • UCR section 5.2.12.8.2.9 • UCR section 5.2.12.8.2.10 • DISA Memo Reference (h)

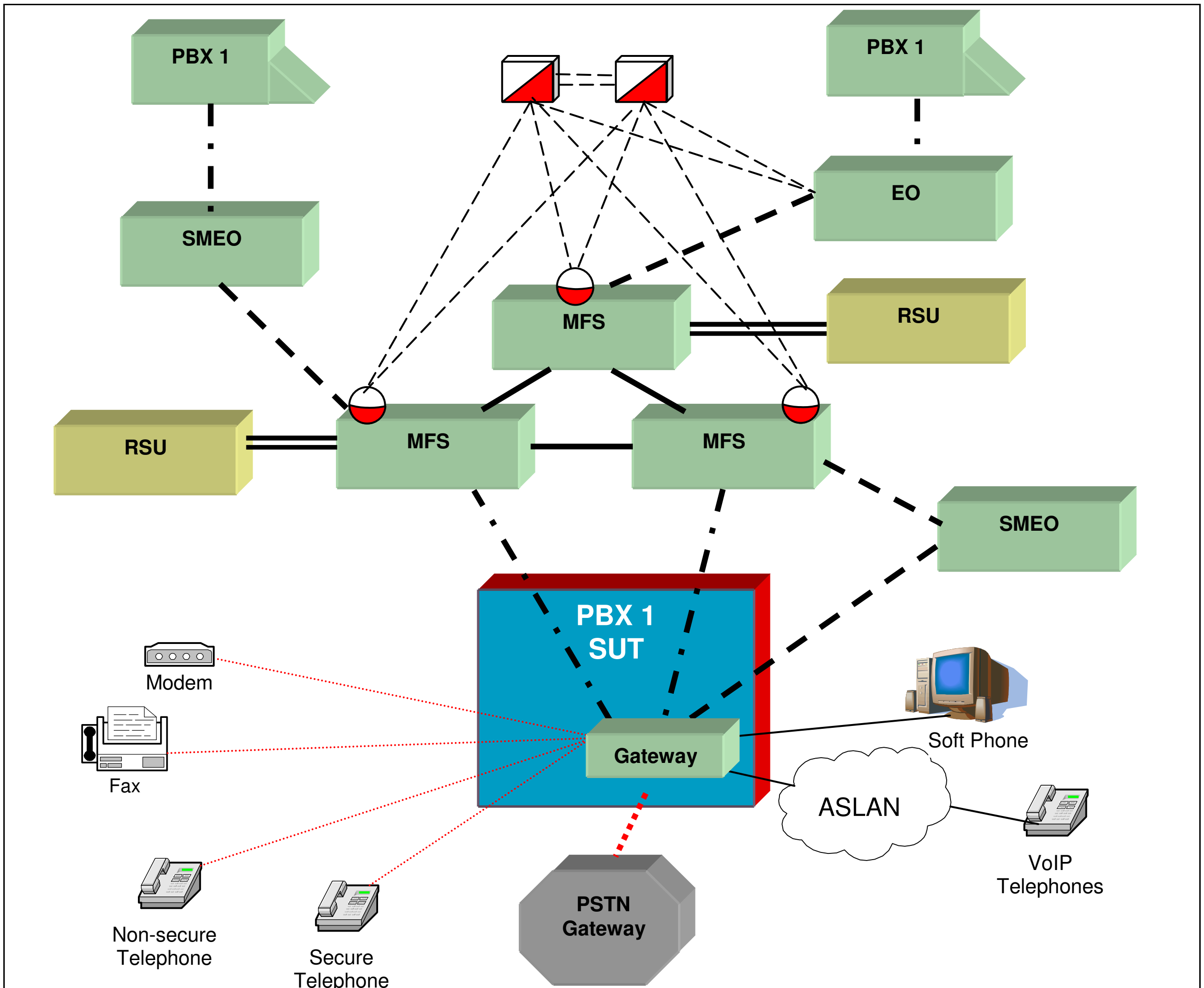
Network Gateways				
Gateway	**Critical**	**Requirements Required or Conditional**		**References**
PSTN (See note.)	No	Trunking	• Positive Identification Control (C) • On-Netting (C) • Off-Netting (C) • Ground Start Line (R) • Immediate Start (C) • Delay Dial (C)	• CJCSI 6215.01C • CJCSI 6215.01C • CJCSI 6215.01C • UCR Section 5.2.4.2.2 • UCR Section 5.2.4.3.2 • UCR Section 5.2.4.3.4
NOTE: Voice, facsimile, data, and VTC service requirements for PSTN are identical to DSN with the exception of MLPP.				

Table 2-1. PBX 1 Requirements (continued)

LEGEND:

802.3u	Standard for carrier sense multiple access with collision detection at 100 Mbps
ANSI	American National Standards Institute
BER	Bit Error Ratio
BRI	Basic Rate Interface
C	Conditional
CAS	Channel Associated Signaling
CJCSI	Chairman of the Joint Chiefs of Staff Instruction
CODEC	Coder/Decoder
DIACAP	DoD Information Assurance Certification and Accreditation Process
DISA	Defense Information Systems Agency
DISR	DoD IT Standards Registry
DoD	Department of Defense
DoDI	Department of Defense Instruction
DP	Dial Pulse
DS0	Digital Signal Level 0 (64 kbps)
DS1	Digital Signal Level 1 (1.544 Mbps) (2.048 Mbps European)
DSN	Defense Switched Network
DTMF	Dual Tone Multi-Frequency
E&M	Ear and Mouth
E1	European Basic Multiplex Rate (2.048 Mbps)
EKTS	Electronic Key Telephone System
FTR	Federal Telecommunications Recommendation
FTR 1080B-2002	Video Teleconferencing Services
G.711	PCM of voice frequencies
GR	Generic Requirement
GR-815	Generic Requirements For Network Element/Network System (NE/NS) Security
H.320	Standard for Narrowband VTC
IEEE	Institute of Electrical and Electronics Engineers
IP	Internet Protocol
IPv6	Internet Protocol version 6
ISDN	Integrated Services Digital Network
IT	Information Technology
ITU-T	International Telecommunication Union-Telecommunication Standardization Sector
kbps	kilobits per second
Mbps	Megabits per second
MFR1	Multi-Frequency Recommendation 1
min	minute
MLPP	Multi-Level Precedence and Preemption
MOS	Mean Opinion Score
NI 1/2	National ISDN Standard 1 or 2
NX56	Data format restricted to multiples of 56 kbps
NX64	Data format restricted to multiples of 64 kbps
PBX	Private Branch Exchange
PBX 1	Private Branch Exchange 1
PCM	Pulse Code Modulation
PCM-24	Pulse Code Modulation - 24 Channels
PCM-30	Pulse Code Modulation - 30 Channels
PRI	Primary Rate Interface
PSTN	Public Switched Telephone Network
Q.955.3	ISDN Signaling Standard for E1 MLPP
R	Required
S/T	ISDN BRI four-wire interface
SS7	Signaling System 7
STE	Secure Terminal Equipment
STIGs	Security Technical Implementation Guides
STU-III	Secure Telephone Unit - 3rd generation
T.4	Standardization of Group 3 facsimile terminals for document transmission
T1	Digital Transmission Link Level 1 (1.544 Mbps)
T1.619a	SS7 and ISDN MLPP Signaling Standard for T1
UCR	Unified Capabilities Requirements
UPS	Uninterruptible Power Supply
VBD	Variable bit data
VoIP	Voice over Internet Protocol
VTC	Video Teleconferencing
yr	year

8. TEST NETWORK DESCRIPTION. The SUT was tested at JITC's Global Information Grid Network Test Facility in a manner and configuration similar to that of the DSN operational environment. Testing of the system's required functions and features was conducted using the notional test configuration depicted in Figure 2-2. The SUT test configuration with an Assured Services Local Area Network (ASLAN) is depicted in Figure 2-3. The SUT was tested as the end-point in relation to the other switches.



LEGEND:

ASLAN	Assured Services Local Area Network
BRI	Basic Rate Interface
CAS	Channel Associated Signaling
DSN	Defense Switched Network
E1	European Basic Multiplex Rate (2.048 Mbps)
EO	End Office
FAX	facsimile
ISDN	Integrated Services Digital Network
Mbps	Megabits per second
MFS	Multifunction Switch
PBX	Private Branch Exchange
PBX 1	Private Branch Exchange Type 1
PRI	Primary Rate Interface
PSTN	Public Switched Telephone Network
RSU	Remote Switching Unit
SMEO	Small End Office
SS7	Signaling System 7
SUT	System Under Test
T1	Digital Transmission Link Level 1 (1.544 Mbps)
TCP/IP	Transmission Control Protocol/Internet Protocol
VoIP	Voice over Internet Protocol

SS7 Service Switching Point (SSP)

SS7 Signal Transfer Point (STP)

PSTN Gateway Trunk

DSN Interswitch Trunk (T1/E1 SS7, T1/E1 CAS, T1/E1 ISDN PRI)

DSN Line (2 Wire Analog)

SS7 Links (A-Link, B-Link, or C-Link)

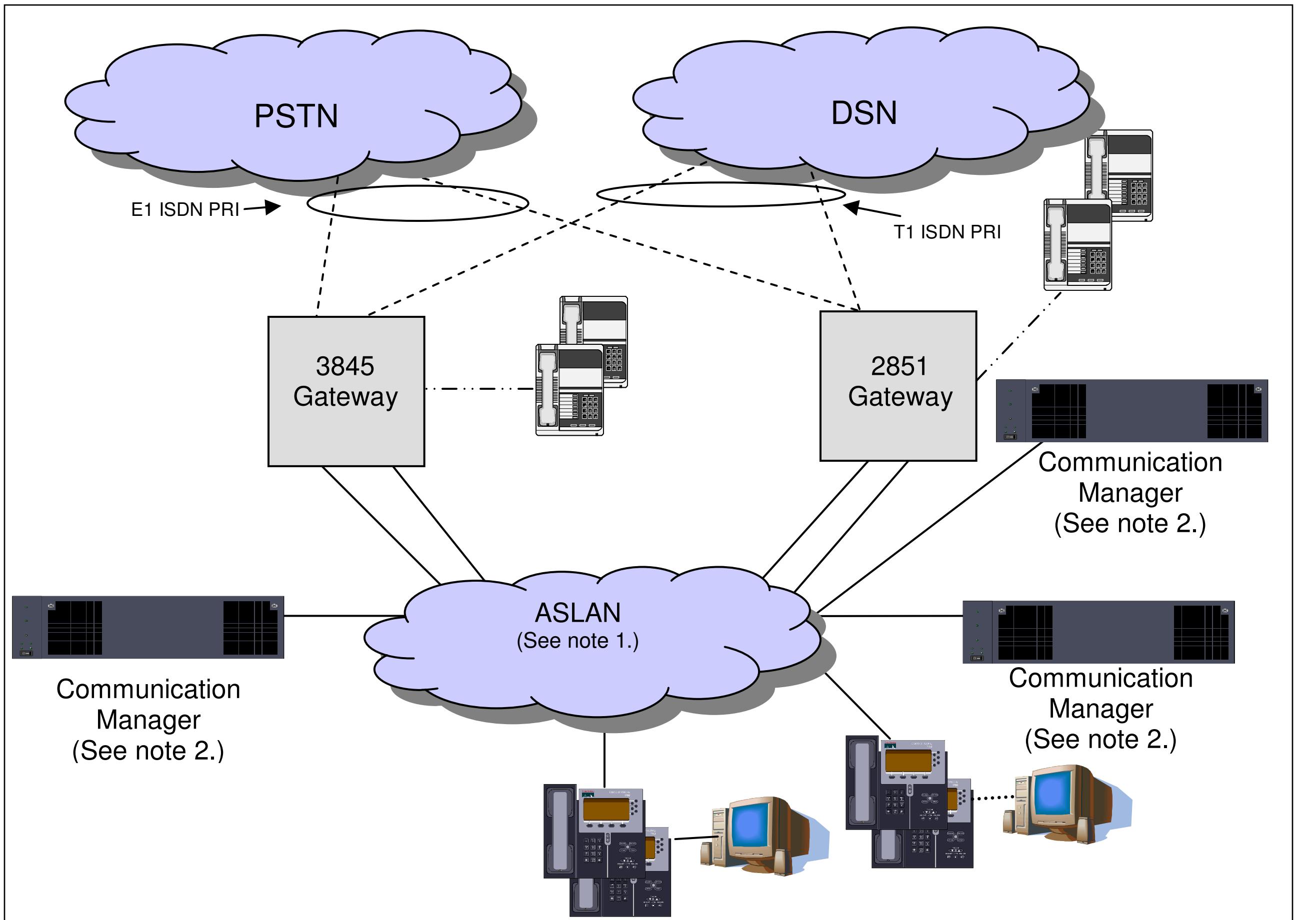
TCP/IP

DSN End Office Access Trunk (T1/E1 ISDN PRI)

DSN PBX Access Trunk (T1/E1 ISDN PRI)

RSU-Host Umbilical Link

Figure 2-2. SUT Notional Test Configuration



NOTES:

1 The SUT is certified with any ASLAN or combination of certified ASLAN components listed on the Unified Capabilities Approved Products List.

2 The following Communication Managers were tested, MCS7835I2, MCS7835H2, MCS7825H3, and MCS7825H4. The other family series of servers which include: MCS7835H1, MCS7845H1, MCS7845H2, MCS7825I4, MCS7835I1, MCS7845I1, MCS7845I2 utilize the same software and similar hardware and JITC analysis determined them to be functionally identical for interoperability certification purposes and they are also certified for joint use.

3 Refer to paragraph 11.a.(5)(a)10.c. for certified shared access rates.

LEGEND:

ASLAN	Assured Services Local Area Network
CP	Cisco Phone
DSN	Defense Switched Network
E1	European Basic Multiplex Rate (2.048 Mbps)
IOS	Internetwork Operating System
ISDN	Integrated Services Digital Network
JITC	Joint Interoperability Test Command
Mbps	Megabits per second
MCS	Media Convergence Server
PBX 1	Private Branch Exchange 1
PC	Personal Computer
PRI	Primary Rate Interface
PSTN	Public Switched Telephone Network
SUT	System Under Test
T1	Digital Transmission Link Level 1 (1.544 Mbps)
TDM	Time Division Multiplexing
VoIP	Voice over Internet Protocol

- - - - - - - - - - - - - - 4-Wire Digital TDM Interfaces

100 Mbps Ethernet

2-Wire Analog

1 Gigabit Ethernet

CP79xx VoIP Telephones (See note 4.)

Analog Telephones

PC for shared access/ PC for IP Communicator

Figure 2-3. SUT Test Configuration with ASLAN

9. SYSTEM CONFIGURATIONS. Table 2-2 provides the system configurations, hardware and software components tested with the SUT. The SUT was tested in an operationally realistic environment to determine interoperability with a complement of DSN switches noted in Table 2-2. Table 2-2 lists the DSN switches which depict the tested configuration and is not intended to identify the only switches that are certified with the SUT. The SUT is certified with switching systems listed on the Unified Capabilities (UC) Approved Products List (APL) that offer the same certified interfaces.

Table 2-2. Tested System Configurations

<table>
<tr><th colspan="2">System Name</th><th colspan="2">Software Release</th></tr>
<tr><td colspan="2">Avaya CS2100</td><td colspan="2">Succession Enterprise (SE) 09.1</td></tr>
<tr><td colspan="2">Siemens EWSD</td><td colspan="2">19d with Patch Set 46</td></tr>
<tr><td colspan="2">Redcom HDX</td><td colspan="2">V.3.0a (R3P0)</td></tr>
<tr><td colspan="2">Avaya S8720</td><td colspan="2">Communication Manager (CM) 4.0 (R014x.00.2.731.7: Super Patch 14419)</td></tr>
<tr><td colspan="2">Cisco MeetingPlace Express</td><td colspan="2">2.1</td></tr>
<tr><th colspan="4">Cisco Unified Communications Manager Version 7.1(2), with IOS Software Release 12.4(22)T2</th></tr>
<tr><th>Component (See note 1.)</th><th>Release</th><th>Sub-component (See note 1.)</th><th>Function</th></tr>
<tr><td>Communication Managers
MCS7835I2, MCS7835H2, MCS7825H3, MCS7825H4, MCS7835H1, MCS7845H1, MCS7845H2, MCS7825I4, MCS7835I1, MCS7845I1, MCS7845I2</td><td>7.1(2.30005.2)</td><td>Not Applicable</td><td>Processing/Signaling</td></tr>
<tr><td rowspan="11">Cisco 3845, 3825 Integrated Services Router (Gateway)</td><td rowspan="11">IOS 12.4(22) T2</td><td>NM HDV2</td><td>TDM Interface NM, HD Voice, 2-slot IP communications enhanced voice/fax</td></tr>
<tr><td>VWIC2 2MFT T1/E1</td><td>Second Generation Voice/WAN Interface Card 2-port RJ-48, Multiflex Trunk T1/E1 (See note 2.)</td></tr>
<tr><td>NM HDV2 2T1/E1</td><td>2-port T1/E1 IP Communications HD voice/fax NM 2 T1/E1 controllers (See note 2.)</td></tr>
<tr><td>NM HDV2 1T1/E1</td><td>1-port T1/E1 IP Communications HD voice/fax NM 2 T1/E1 controllers (See note 2.)</td></tr>
<tr><td>VIC3 4FXS/DID</td><td>Voice Interface Card, 4-port, RJ-11, Foreign Exchange Station, DID</td></tr>
<tr><td>VIC3 2FXS</td><td>Voice Interface Card, 2-port, Foreign Exchange Station</td></tr>
<tr><td>VIC 4FXS/DID</td><td>Voice Interface Card, 4-port, RJ-11, Foreign Exchange Station, DID</td></tr>
<tr><td>VIC2 2FXS</td><td>Voice Interface Card, 2-port, RJ-11, Foreign Exchange Station</td></tr>
<tr><td>EM HDA 8FXS</td><td>8-port analog Foreign Exchange Station expansion module for voice and fax (See note 3.)</td></tr>
<tr><td>EVM HD 8FXS/DID</td><td>HD analog and digital extension module for voice and fax</td></tr>
<tr><td>EM3 HDA 8FXS/DID</td><td>8-Port HD analog and digital extension module for voice and fax (See note 3.)</td></tr>
</table>

Table 2-2. Tested System Configurations (continued)

Cisco Unified Communications Manager Version 7.1(2) with IOS Software Release 12.4(22)T2 (continued)			
Component (See note 1.)	**Release**	**Sub-component (See note 1.)**	**Function**
Cisco 2851, 2821, 2811 Integrated Services Router (Gateway)	IOS 12.4(22)T2	**NM HD 2VE**	2-slot IP communications enhanced voice/fax network module
		VIC 4FXS/DID	Voice Interface Card, 4-port, RJ-11, Foreign Exchange Station, DID
		VWIC2 2MFT T1/E1	Second Generation Voice/WAN Interface Card 2-port RJ-48, Multiflex Trunk T1/E1 (See note 2.)
		EVM HD 8FXS/DID	HD analog and digital extension module for voice and fax
		EM HDA 8FXS	8-port analog Foreign Exchange Station expansion module for voice and fax (See note 3.)
		NM HDV2 2T1/E1	2-port T1/E1 IP Communications HD voice/fax NM, 2 T1/E1 controllers (See note 2.)
		NM HDV2 1T1/E1	1-port T1/E1 IP Communications HD voice/fax NM, 1 T1/E1 controllers (See note 3.)
		VWIC2 1MFT T1/E1	Second Generation Voice/WAN Interface Card 1-port RJ-48, Multiflex Trunk T1/E1 (See note 2.)
		VIC3 4FXS/DID	Voice interface card, 4-port, RJ-11, foreign exchange station, DID
		EM3 HDA 8FXS/DID	8-port HD analog and digital extension module for voice and fax (See note 3.)
		VIC3 2FXS	Voice Interface card, 2-port, RJ-11, Foreign exchange station
		VIC 4FXS/DID	Voice Interface Card, 4-port, RJ-11, Foreign Exchange Station, DID
		VIC2 2FXS	Voice Interface Card, 2-port, RJ-11, Foreign Exchange Station
CP-7940G and CP-7960G (See note 4.)	P00308010100	Not Applicable	IP Phone (with push-to-talk handset or with standard handset)
CP-7970G and CP-7971G	SCCP70.8-5-2S	Not Applicable	IP Phone (with push-to-talk handset or with standard handset)
CP-7931G	SCCP31.8-5-2S	Not Applicable	IP Phone (with push to talk handset or with standard handset)
CP-7911G and 7906G	SCCP11.8-5-2S	Not Applicable	IP Phone (with push-to-talk handset or with standard handset)
CP-7941G, CP-7941G-GE, CP-7961G, and CP-7961G-GE	SCCP41.8-5-2S	Not Applicable	IP Phone (with push-to-talk handset or with standard handset)
CP-7942G and CP-7962G	SCCP42.8-5-2S	Not Applicable	IP Phone (with push-to-talk handset or with standard handset)
CP-7945G and CP-7965G	SCCP45.8-5-2S	Not Applicable	IP Phone (with push-to-talk handset or with standard handset)
CP-7975G	SCCP75.8-5-2S	Not Applicable	IP Phone (with push-to-talk handset or with standard handset)
7914	Load: S00105000400	Not Applicable	Expansion module
7915	B015-1-0-3	Not Applicable	Expansion module
7916	B015-1-0-3	Not Applicable	Expansion module
General Dynamics C4 Systems Sectéra® vIPer™ (See note 5.)	Release 1.0, Software ver.6.04	Not Applicable	IP Phone (with standard handset)
Telecore 2151	2AE-00056-0003	Not Applicable	IP Phone (with push-to-talk handset or with standard handset), 100 Mbps shared access[6]
CIS Secure DTD-7961-T-SG-SC-SC-X-X (See note 7.)	SCCP41.8-5-2S	Not Applicable	7961G TEMPEST version with 100 Mbps SC Fiber LAN and PC interfaces, TSG Positive Disconnect, no speakerphone, shared access

Table 2-2. Tested System Configurations (continued)

Cisco Unified Communications Manager Version 7.1(2) with IOS Software Release 12.4(22)T2 (continued)			
Component (See note 1.)	**Release**	**Sub-component (See note 1.)**	**Function**
CIS Secure DTD-7975-X-XSC-RJ-ME-SE (See note 7.)	SCCP75.8-5-2S	Not Applicable	7975G Standard with 1000 Mbps SC Fiber LAN and RJ45 PC interfaces, shared access
CRYPTEK CT915-V-P1-003 (See note 7.)	SCCP41.8-5-2S	Not Applicable	7961G IP phone, Fiber TEMPEST version with 100MB Fiber LAN and no shared access
Walker WS-2620	Not Applicable	Not Applicable	Push to Talk Handset for Cisco 7900 Series phones
Cisco IP Communicator (See note 8.)	7.0.5	Not Applicable	Cisco Softphone Application

NOTES:

1 Components bolded and underlined were tested by JITC. The other components in the family series were not tested; however, they utilize the same software and similar hardware and JITC analysis determined them to be functionally identical for interoperability certification purposes and they are also certified for joint use.

2 These components are certified in the DSN with T1 ISDN PRI interface. These components are certified in the PSTN with the T1 ISDN PRI and E1 ISDN PRI interfaces.

3 The EM HDA 8FXS and EM3 HDA 8FXS/DID expansion modules require the EVM HD module. Up to two EM HDA 8FXS or EM3 HDA 8FXS/DID expansion modules are supported for each EVM HD.

4 The SUT met all IPv6 requirements through testing and LoC with the following exception: The Cisco CP-7940G and CP-7960G end instruments did not meet dual stack IPv6 requirements. These end instruments represent legacy end instruments which are IPv4 only; however, the SUT met the minimum requirement for dual stack IPv6 end instruments with the other IP end instruments listed in this table and a dual stack call control agent in accordance with the interim UCR IPv6 rules of engagement, Reference (g).

5 This instrument is certified specifically with 2800 and 3800 series gateways with IOS 12.4(22) T2 or higher version listed on the UC APL.

6 Although the Telecore 2151 supports both 100 Mbps and 1 Gbps shared access, due to MOS scores below the required 4.0 for 1 Gbps shared access, the Telecore 2151 is only certified for shared access at 100 Mbps.

7 CIS Secure Computing and Cryptek add security modifications which are physical in nature and do not affect software. The CIS Secure Computing and Cryptek security modifications can be applied to any certified Cisco IP phone.

8 Reference (h) is a DISA memo that stipulates interim softphone requirements that supersede the current UCR 2008 requirements until they are implemented in Change 1. The softphone shall be functionally identical to a traditional IP “Hard” telephone and will be required to provide voice features and functionality provided by a traditional IP “Hard” Telephone with following exceptions:

a. Audible and visual alerting to the end user of an incoming call, even if the application is running in the background.
b. Softphone application shall be exempt from reliability, availability and performance (packet loss, jitter, latency) requirements.
c. Microphone and speaker or headphone, or any other audio input/output device, Ethernet interface(s), and mouse (point and click) interaction.
d. IPv6 is not required.

LEGEND:

APL	Approved Product List
CP	Cisco Phone
CS	Communication Server
DID	Direct Inward Dialing
DISA	Defense Information Systems Agency
DSN	Defense Switched Network
E1	European Basic Multiplex Rate (2.048 Mbps)
EM	Expansion Module
EVM	Extension Voice Module
EWSD	Elektronisches Wählsystem Digital
Fax	facsimile
FXS	Foreign Exchange Station
G	10/100BaseT Ethernet (A Cisco part designator on their IP phone.)
Gbps	Gigabits per second
GE	Gigabit Ethernet (A Cisco part designator on their IP phone.)
HD	High Density
HDA	High Density Analog
HDX	High Density Exchange
IOS	Internetwork Operating System
IP	Internet Protocol
Ipv4	Internet Protocol version 4
IPv6	Internet Protocol version 6
ISDN	Integrated Services Digital Network
JITC	Joint Interoperability Test Command
LAN	Local Area Network
Mbps	Megabits per second
MCS	Media Convergence Server
MFT	Multiflex Trunk
MOS	Mean Opinion Score
NM	Network Module
PC	Personal Computer
PRI	Primary Rate Interface
PSTN	Public Switched Telephone Network
RJ	Registered Jack
SC	fiber connector (square push-in)
SCCP	Skinny Call Control Protocol
SUT	System Under Test
T1	Digital Transmission Link Level 1 (1.544 Mbps)
TDM	Time Division Multiplexing
UC	Unified Capabilities
UCR	Unified Capabilities Requirements
V	Voice
VE	Voice/Fax Enhanced
VIC	Voice Interface Card
VWIC	Voice WAN Interface Card
WAN	Wide Area Network

10. TESTING LIMITATIONS. None.

11. TEST RESULTS

a. Discussion

(1) DSN Trunk Interfaces

(a) The SUT met all critical CRs and FRs for T1 ISDN PRI NI 1/2 ANSI T1.619a interface with one minor exception. The SUT does not support Non Facility Associated Signaling (NFAS) on their T1 ISDN PRI NI2 interface in accordance with the UCR. This discrepancy was adjudicated by DISA as having a minor operational impact for a PBX 1 and DISA has stated their intent to modify this requirement as a conditional requirement for a PBX1 in the next update (Change 1) to the UCR.

1. E1 ISDN PRI is supported by the SUT; however, it does not support ITU-T Q.955.3 MLPP. Therefore, this interface is not certified as a DSN interface. This is not a required DSN interface for a PBX 1.

2. The T1 Channel Associated Signaling (CAS) interface is supported by the SUT; however, the T1 CAS interface is not certified due to the following critical discrepancies: The SUT T1 CAS interface does not recognize Remove from Service (Busy Out) or Restore to Service (Make Idle) condition from the distant end switch. In addition, when the Busy Out condition is invoked across the T1 CAS interface, it causes the SUT 3845 and 2851gateway T1 CAS interface to deregister from its current subscriber and reregister to an alternate subscriber and then within 1 to 5 minutes repeat the process and go back to its original subscriber. During this transition period, calls are unable to process to the SUT. The T1 CAS interface is therefore not certified for use within the DSN. There is no operational impact because the T1 CAS interface is not a required interface for a PBX 1.

(2) DSN Line Interfaces. The SUT met all critical interoperability certification requirements for 2-Wire Loop Start Analog (GR-506-CORE) and Voice over Internet Protocol (VoIP) DSN line interfaces with the minor exceptions listed in paragraphs 11.a.(3)(g)1 and 11.a.(5)(a)8.

(3) Features and Capabilities

(a) Common Features. The SUT met all critical interoperability certification requirements for Features and Capabilities with the following exceptions: Call Forward No Answer, Call Forward Busy, and Multi-Line Hunt Service are supported on both VoIP and analog phones. Call Forward Variable, Three-way Calling, Call Hold, Call Pick-up, and Call Transfer are supported on VoIP stations only. These features are required for a PBX 1 for all instruments; however, since this requirement is a new UCR requirement and the vendor has 18 months to develop it (July 2010), the operational impact is minor. Since the SUT test window started before the 18 month development window expired, DISA stated this new feature requirement does not apply. All of the features on the VoIP phones were tested using multiple line appearances. A minimum of two line appearances is required to meet the MLPP interoperability requirements for Call Features with the exception of call hold, call pickup, and call forwarding functions. Although the SUT does not support Precedence Call Waiting, they do support multiple call appearances on their VoIP phones. This provides the ability for a user to receive additional calls while active with another call. A short "ping" ring is not provided when calls are forwarded; however, the phone does visually display that call forward variable is enabled. There is a minor operational impact. When a ROUTINE call is placed to a hunt group and a ring-no-answer condition occurs, the calling party is diverted to the MLPP alternate directory number. This configuration must be done to allow correct treatment to be provided to precedence calls above ROUTINE that are placed to the hunt group. There is no operational impact. Refer to Table 2-3 for a list of the Common Features supported for the phone types and associated test results.

Table 2-3. SUT Common Call Feature Availability

Call Feature	Phone Type	
	Analog	IP[1]
Precedence Call Waiting	Not Supported[2]	Not Supported[3]
Call Hold	Not Supported[2]	Passed
Call Forwarding No Answer	Passed	Passed
Call Forwarding Busy	Passed	Passed
Call Forwarding Variable	Not Supported[2]	Passed[4]
Three-Way Calling	Not Supported[2]	Passed[5]
Call Transfer	Not Supported[2]	Passed
Multi-line Hunt Service	Passed[5]	Passed[5]
Call Pickup	Not Supported	Passed

NOTES:

1 All of the features on the VoIP phones were tested using multiple line appearances. A minimum of two line appearances is required to meet the MLPP interoperability requirements for Call Features with the exception of call hold, call pickup, and call forwarding functions. There is no operational impact.

2 The SUT analog gateway does not support the following required line features: Call Transfer, Call Hold, Precedence Call Waiting, Call Forwarding Variable, Three-Way Calling, and Call Pickup. These features are required for a PBX 1 for all instruments; however, since this requirement is a new UCR requirement and the vendor has 18 months to develop it (July 2010), the operational impact is minor. Since the SUT test window started before the 18 month development window expired, DISA stated this new feature requirement does not apply. The operational impact is minor.

3 Although the SUT does not support Precedence Call Waiting, they do support multiple call appearances on their VoIP phones. This provides the ability for a user to receive additional calls while active with another call. There is no operational impact.

4 A short "ping" ring is not provided when calls are forwarded; however, the phone does visually display that call forward variable is enabled. There is a minor operational impact.

5 When a ROUTINE call is placed to a hunt group, and a ring-no-answer condition occurs, the calling party is diverted to the MLPP alternate directory number. This configuration must be done to allow correct treatment to be provided to precedence calls above ROUTINE that are placed to the hunt group. There is no operational impact.

LEGEND:

DISA	Defense Information Systems Agency	SUT	System Under Test
IP	Internet Protocol	UCR	Unified Capabilities Requirements
MLPP	Multi-Level Precedence and Preemption	VoIP	Voice over Internet Protocol
PBX 1	Private Branch Exchange 1		

(b) Attendant. This feature is not supported by the SUT. This is not a required feature for a PBX 1. There is no risk associated with the SUT not supporting this feature.

(c) Public Safety. The SUT meets the minimum critical interoperability requirements for Public Safety which is basic emergency service 911 service. This feature allows the user to dial 911 and the SUT then retranslates it to be routed to a Public Safety Answering Point via a trunk or line. The following public safety features are not supported and therefore are not covered in this certification: Trace of terminating calls, Outgoing call trace, Tandem call trace, and Trace of a call in progress. There is no operational impact because these public safety features are not required for a PBX 1.

(d) Conferencing. Meet-Me Conferencing can be met through the use of an optional adjunct conferencing system called the Cisco Meeting Place Express which is covered under a separate certification. The SUT does not support Preset Conferencing or Progressive Conferencing. These features are not required for a PBX 1. There is no risk associated with the SUT not supporting these features.

(e) Nailed-up Connections. This feature is not supported by the SUT. This is not a required feature for a PBX 1. There is no risk associated with the SUT not supporting this feature.

(f) DSN Hotline Services. This feature is not supported by the SUT. This is not a required feature for a PBX 1. There is no risk associated with the SUT not supporting this feature.

(g) Multi-Level Precedence and Preemption (MLPP). Met all critical CRs and FRs with the following minor exceptions:

1. The SUT does not support an MLPP global diversion number. Each station must be individually configured with a precedence diversion number from a single location using the Bulk Administration Tool provided with the Cisco Unified Communication Manager. The operational impact is minor because diversion settings can be configured for all of the stations provisioned on the switch from a single location.

2. The SUT does not support the Loss of Command and Control (C2) announcement. This announcement is invoked only when a DSN subscriber is automatically routed to a non-MLPP network. This anomaly was adjudicated as minor because this announcement would rarely be invoked on a PBX 1. Furthermore, DISA, in coordination with the Joint Staff, stated their intent to modify the next update of the UCR to change the Loss of C2 announcement from required to conditional for a PBX 1.

(h) Call Processing. Met all critical CRs and FRs.

(i) ISDN Services. Met all critical CRs and FRs. The SUT does not support NFAS on their ISDN PRI NI2 interface. DISA's adjudication of this discrepancy was completed on 17 December 2008 and was ruled to have a minor operational impact. Furthermore, DISA stated they intent to modify the next update of the UCR to change NFAS for a PBX 1 from required to conditional.

(j) Synchronization. All critical interoperability certification CRs and FRs were met for this feature by the SUT. The SUT supports line timing mode and Internal Stratum 4 for synchronization.

(k) Reliability. All critical interoperability certification CRs and FRs for this feature were met by the SUT and met by vendor LoC.

(l) Security. Security is tested by DISA-led Information Assurance test teams and published in a separate report, Reference (c).

(4) Network Gateways. The SUT met all critical interoperability certification requirements for the Public Switched Telephone Network (PSTN) Network Gateway. The interfaces certified for the PSTN are T1 ISDN PRI NI 1/2 (ANSI T1.607), ITU-T Q.931 E1 ISDN PRI, and 2-Wire Analog Ground Start Line (GR-506 CORE). The SUT offers a T1 CAS trunk interface; however, it was not certified. Critical interoperability discrepancies (refer to paragraph 11.a(1)(a)2.) were discovered during testing. The SUT T1 CAS interface is not certified for use within the DSN. This is not a required interface for a PBX 1.

(5) VoIP. The SUT is certified with any ASLAN or any combination of certified ASLAN components listed on the UC APL.

(a) VoIP System. The UCR, section 5.2.12.8.2, outlines the requirements for the VoIP system. The VoIP system requirements encompass end-to-end VoIP requirements. The following paragraphs detail the results of the SUT VoIP solution.

1. Voice Quality. In accordance with the UCR, section 5.2.12.8.2.1, VoIP calls shall have an average Mean Opinion Score (MOS) of at least 4.0 as measured in accordance with ITU-T P.800 voice quality standards. This applies from handset to handset and for intra- and inter-switch calls end-to-end. The SUT meets MOS requirements with an average of 4.38 for 118 test calls. The SUT met this requirement with all VoIP phones to include the Cisco IP Communicator softphone.

2. Codec. In accordance with the UCR, section 5.2.12.8.2.2, the International Telecommunication Union - Telecommunication Standardization Sector (ITU-T) G.711 Pulse Code Modulation (PCM) CODEC with a 20 ms packet fill was required and was met by the SUT VoIP solution.

3. MLPP. In accordance with the UCR, section 5.2.12.8.2.3, the VoIP system shall meet all MLPP requirements identified in UCR, section 3. All critical MLPP features and functions were met.

4. Security. Security requirements in accordance with the UCR, section 5.2.12.8.2.4, are verified using the Information Assurance Test Plan. Results of

the security testing are reported in a separate test report generated by the DISA Information Assurance test personnel, Reference (c).

5. Network Management (NM). In accordance with the UCR, section 5.2.12.8.2.5, the vendor is required to provide a management system to monitor the performance of the ASLAN portion of the VoIP system. This requirement was covered under a separate certification for the respective ASLANs listed on the UC APL. In accordance with the UCR, section 5.3.8, the switching system NM requirements are not required for a PBX 1 and were not tested.

6. Synchronization. In accordance with the UCR, section 5.2.10.1.1.2, the SUT is required to derive timing with line timing mode and an internal clock of stratum 4 or better. The SUT met this requirement.

7. Latency. The UCR, section 5.2.12.8.2.7, states that one-way system latency for the VoIP system must be 60 milliseconds (ms) or less as averaged over any five-minute period. The latency requirement is measured from IP or analog handset to the egress trunk. The SUT latency measurements were conducted from each phone type supported by the SUT for IPv4 and IPv6 traffic. Over 80, 20-minute interswitch phone calls were measured with a latency between 46 ms to 56 ms, with an average of 51 ms.

8. Internet Protocol version 6 (IPv6). In accordance with UCR, section 5.3.5, all systems submitted for testing must be IPv6 capable. Dual Stack solutions are preferred and tunneling solutions are unacceptable. IPv6-capable products, in accordance with UCR, section 4.3.1.3, can create or receive, process, and send or forward IPv6 packets in mixed IPv4/v6 environments. IPv6-capable networks can receive, process, and forward IPv6 packets from/to devices within the same network and from/to other networks and systems, where those networks and systems may be operating with only IPv4, only IPv6, or both IPv4 and IPv6. IPv6 capable products shall:

a. Conform to the requirements of the DoD IPv6 Standard Profiles for IPv6 Capable Products document contained in the DISR. This requirement was met with an LoC submitted by the vendor.

b. Possess a migration path and written commitment to upgrade by the company Vice President or equivalent, as the IPv6 standard evolves. This requirement was met with a LoC submitted by the vendor.

c. Ensure IPv6 technical support is available. This requirement was met with an LoC submitted by the vendor.

d. Conform to National Security Agency (NSA) and/or Unified Cross Domain Management Office requirements for Information Assurance products.

All of the SUT components covered under this certification met the IPv6 criteria through testing and the LoC with the following exceptions, which were adjudicated by DISA on 2 September 2009 as having a minor operational impact:

a. The Cisco CP-7940G and CP-7960G end instruments did not meet dual stack IPv6 requirements. These end instruments represent legacy end instruments which are IPv4 only; however, the SUT met the minimum requirement for dual stack IPv6 end instruments with the other end instruments listed in Table 2-2 and a dual stack call control agent in accordance with the interim UCR IPv6 Rules of Engagement signed by OSD on 31 July 2009, Reference (g).

b. During initial boot up of the CP-7940G and CP-7960G phones, some of the User Datagram Protocol (UDP)/Trivial File Transfer Protocol (TFTP) traffic has a Differentiated Services Code Point (DSCP) value of 4 and 802.1Q value of 5 and can not be changed.

c. The SUT management workstation provided during testing did not assign DSCP values for Operational Administration and Maintenance (OAM) IP traffic.

d. The IP phones are incorrectly tagging IPv6 Transmission Control Protocol (TCP) traffic during power up.

e. The soft Client is incorrectly tagging all traffic during power up.

f. The 802.1Q COS tag values are not independently configurable from the DSCP values.

g. The MCS7825H4 Communication Manager server stops transmitting IP Traffic if the Network Interface Card (NIC) failover is enabled. The NIC failover is offered on this server but is not required for a PBX 1. This setting will be annotated in the deployment guide for this server.

h. End Instruments, except for the Telecore 2151, do not support the manual configuration of the IPv6 default gateway.

i. The 2851 and 3845 gateways cannot set IPv6 flow label value to zero for Real-time Transport Protocol (RTP) media traffic.

j. Communication Managers are incorrectly tagging UDP/TFTP traffic to the end instrument. IPv6 Traffic Class and IPv4 DSCP values for signaling cannot be set to the full range of 0-63 in accordance with the UCR. The SUT can only tag Traffic Class and DSCP values for signaling with the following values: 0, 8, 12, 12, 14, 16, 18, 20, 22, 24, 26, 28, 30, 32, 34, 36, 40, 46, 48, and 56.

9. In accordance with the UCR, section 5.2.12.8.2.9, the VoIP system (i.e. Media Gateway and Session Control Agent) shall meet the following requirements:

a. All components shall be capable of implementing Service Class tagging using the 8-bit Traffic Class in the IPv6 header and DSCP field in the IPv4 header. The SUT meets the requirement.

b. All session control components shall be capable of assigning DSCP (0-63) to any distinct service class for traffic that traverses the device in accordance with UCR, Table 5.3.1-3. The SUT session control components can only have the signaling service class traffic configured for 21 different DSCP values and not the full range required. The Traffic Class and DSCP values for media can be assigned to any value from 0-63. The MCS7835 and the MCS7825 OAM traffic is tagged at zero and is not configurable. In addition, the 2851 and 3845 gateways are tagging IPv4 RTP Control Protocol (RTCP) traffic at zero and it is not configurable. These discrepancies were adjudicated by DISA as having a minor operational impact.

c. For VoIP, video, and data end products, any end system that supports convergence must preassign the VLAN using Institute of Electrical and Electronics Engineers (IEEE) 802.1Q tags prior to the frames entering the ASLAN in accordance with UCR, section 5.3.1.7.4. For end-systems that support just one media, the LAN can assign the VLAN based on port-based VLAN assignment. The SUT CallManager does not support more than one media; therefore, VLAN tagging is not supported. There is no operational impact.

10. In accordance with the UCR, section 5.2.12.8.2.9, the VoIP system end user devices shall meet the following requirements:

a. All end instruments shall be capable of implementing Service Class tagging using the 8-bit Traffic Class in the IPv6 header and DSCP field in the IPv4 header. The SUT end instruments that support IPv6 dual stack used class tagging in the respective IP headers for IPv4 and IPv6, which meets the requirement.

b. All end instrument components shall be capable of assigning DSCP (0-63) to any distinct service class for traffic that traverses the device in accordance with UCR, Table 5.3.1-3. The DSCPs may be assigned by either having the end instrument itself assign the Traffic Class and DSCP tag to the distinct service class or having the call control portion of the VoIP system tell the end instrument what distinct service class to assign. The SUT end instruments components only have the ability to configure 21 different DSCP values for signaling service class traffic. The DSCP values for media can be assigned to any DSCD value from 0-63. The DSCP value of traffic on the CP7940 and CP7960 phones is configured to 4, and it cannot be changed. These discrepancies were adjudicated by DISA as having a minor operational impact. A management workstation that meets the requirement for assigning DSCP values in the IPv4 header was not provided for test.

c. For VoIP, video, and data end products, any end system that supports convergence must preassign the VLAN using IEEE 802.1Q tags prior to the frames entering the ASLAN in accordance with UCR, section 5.3.1.7.4. For end-systems that support just one media, the LAN can assign the VLAN based on port-based VLAN assignment. The SUT end instruments have the capability of supporting shared access. Additionally the SUT end instruments have the capability to tag Real Time Traffic with the appropriate VLAN Identifier value. The Cisco VoIP phones that met the critical interoperability requirements for certification with 100 Mbps interface were the: CP7906G, CP7911G, CP7940G, CP7941G, CP7941G-GE, CP7942G, CP7945G, CP7960G, CP7961G-GE, CP7961G, CP7962G, CP7965G, CP7970G, CP7971G-GE, CP7975G, Tempest phone Cryptek 7961G, Tempest phone CIS 7961G, Telecore 2151 and Tempest phone CIS 7975G. The above phones have been tested and are certified for shared access (i.e., same switch port is shared by PC and IP phone) with the exception of the CP7906G. The CP7906G phone does not support shared access. The following phones are also certified for 1 Gbps shared access: CP7971G-GE, CP7975G, CP7965G, CP7945G, CP7941G-GE, CP7961G-GE, and Tempest phone CIS 7975G. The Tempest phones Cryptek 7961G, and CIS 7961G must have "Port Policing" configured at the network interface in order to allow proper port shared access. The CP7970G and CP7971G-GE phones are capable of web browsing; however, this feature was not tested, is not covered by this certification. All VoIP phones were tested using Secure Real Time Protocol (SRTP) which encrypts the media stream. The SRTP is able to encrypt only IP phone to IP phone intra-switch traffic and IP phone to gateway intra-switch traffic. All other calls (i.e. analog to analog, or analog to gateway traffic) are not encrypted.

11. Reference (h) is a DISA memo that stipulates interim softphone requirements that supersede the current UCR 2008 requirements until they are implemented in Change 1. The SUT Cisco IP Communicator met all of the critical requirements in accordance with this memorandum as listed below with minor exceptions which are noted in this summary.

The softphone shall be functionally identical to a traditional IP "Hard" telephone and will be required to provide voice features and functionality provided by a traditional IP "Hard" Telephone with following exceptions:

- Audible and visual alerting to the end user of an incoming call, even if the application is running in the background.

- Softphone application shall be exempt from reliability, availability, and performance (packet loss, jitter, latency) requirements.

- Microphone and speaker or headphone, or any other audio input/output device, Ethernet interface(s), and mouse (point and click) interaction.

- IPv6 is not required.

b. System Interoperability Results. The SUT is certified for joint use in the DSN as a PBX 1 and PBX 2 in accordance with the requirements set forth in the UCR. The identified test discrepancies that remained open after software patches were applied and regression testing was completed have an overall minor operational impact. The following components were not tested; however, they utilize the same software and similar hardware as tested components and JITC analysis determined them to be functionally identical for interoperability certification purposes: MCS7835H1, MCS7845H1, MCS7845H2, MCS7825I4, MCS7835I1, MCS7845I1, MCS7845I2, 3825, 2821and 2811. The SUT interoperability test summary is shown in Table 2-4. The SUT Interoperability Requirements/Status is shown in Table 2-5.

Table 2-4. SUT Interoperability Test Summary

DSN Trunk Interfaces			
Interface & Signaling	**Critical**	**Status**	**Remarks**
T1 CAS (DTMF, MFR1, DP)	No	Not Certified	The SUT T1 CAS interface was tested but did not meet all critical CRs and FRs. The SUT T1 CAS interface is therefore not certified by JITC. This is not a required interface for a PBX 1.[1]
E1 CAS (DTMF, MFR1, DP)	No (Europe only)	Not Tested	E1 CAS is supported by the SUT. However, it was not tested. The SUT E1 CAS interface is therefore not certified by JITC. This is not a required interface for a PBX 1.
T1 ISDN PRI NI 1/2 (ANSI T1.619a)	Yes	Certified	Met all critical CRs and FRs with the following minor exception: The SUT T1 ISDN PRI NI2 interface does not support NFAS.[2]
E1 ISDN PRI (ITU-T Q.955.3)	No (Europe only)	Not Certified	The E1 ISDN PRI interface is supported by the SUT; however, it does not support ITU-T Q.955.3 MLPP. The SUT E1 ISDN PRI interface is therefore not certified by JITC for use within the DSN. This interface is certified only for PSTN. This is not a required DSN interface for a PBX 1.
DSN Line Interfaces			
Interface & Signaling	**Critical**	**Status**	**Remarks**
2-Wire Analog Loop Start (GR-506-CORE)	Yes	Certified	Met all critical CRs and FRs with the following minor exception: The SUT gateway analog interface does not support required line features.[3] The operational impact is minor.
ISDN BRI NI 1/2 (ANSI T1.619a)	No	Not Tested	This interface is not supported by the SUT. This is not a required interface for a PBX 1. There is no risk associated with the SUT not supporting this interface.
2-Wire Proprietary Digital	No	Not Tested	This interface is not supported by the SUT. This is not a required interface for a PBX 1. There is no risk associated with the SUT not supporting this interface.
VoIP (Ethernet IEEE 802.3u)	No	Certified	Met all critical CRs and FRs with the following minor exception: The Cisco CP-7940G and CP-7960G end instruments did not meet dual stack IPv6 requirements.[4]

Table 2-4. SUT Interoperability Test Summary (continued)

DSN Features and Capabilities			
Features and Capabilities	**Critical**	**Status**	**Remarks**
Common Features	Yes	Certified	Met all critical CRs and FRs with the following minor exception: Full compliance of DSN Common Call Features was not met. The operational impact is minor.[3]
Attendant	No	Not Tested	This feature is not supported by the SUT. This is not a required feature for a PBX 1. There is no risk associated with the SUT not supporting this feature.
Public Safety	Yes	Certified	All public safety features are conditional. The SUT met all critical CRs and FRs for Basic 911. The SUT does not support the other public safety features. These are not required features for a PBX 1. There is no risk associated with the SUT not supporting these features.[5]
Conferencing	No	Not Certified	The SUT can support Meet-Me Conferencing through the optional MeetingPlace Express.[6] The SUT does not support Preset Conferencing or Progressive Conferencing. These features are not required for a PBX 1. There is no risk associated with the SUT not supporting these features.
Nailed-up Connections	No	Not Tested	This feature is not supported by the SUT. This is not a required feature for a PBX 1. There is no risk associated with the SUT not supporting this feature.
DSN Hotline Services	No	Not Tested	This feature is not supported by the SUT. This is not a required feature for a PBX 1. There is no risk associated with the SUT not supporting this feature.
MLPP	Yes	Certified	Met all critical CRs and FRs with the following minor exceptions: The SUT does not support a global diversion number.[7] The SUT does not support the Loss of Command and Control announcement.[8]
Call Processing	Yes	Certified	Met all critical CRs and FRs.
ISDN Services	Yes	Not Tested	This feature is not supported by the SUT. This is not a required feature for a PBX 1. There is no risk associated with the SUT not supporting this feature.
Synchronization	Yes	Certified	Met all critical CRs and FRs.
Reliability	Yes	Certified	Met all critical CRs and FRs.
Security	Yes	Certified	See note 9.
VoIP System	No	Certified	The SUT is certified for VoIP specifically with any certified ASLAN or ASLAN components posted on the UC APL. (See notes 4 and 10.)

Network Gateways				
Gateway	**Interface & Signaling**	**Critical**	**Status**	**Remarks**
PSTN	T1 CAS (DTMF, MFR1, DP)	No	Not Certified	SUT T1 CAS interface was tested but did not meet requirements. The SUT T1 CAS interface is therefore not certified by JITC. This is not a required interface for a PBX 1.[1]
	E1 CAS (DTMF, MFR1, DP)	No (Europe only)	Not Tested	E1 CAS is supported by the SUT; however it was not tested. The SUT E1 CAS interface is therefore not certified by JITC. This is not a required interface for a PBX 1.
	T1 ISDN PRI NI 1/2 (ANSI T1.607)	No	Certified	Met all critical CRs and FRs with the following minor exception: The SUT T1 ISDN PRI NI/2 interface does not support NFAS.[2]
	E1 ISDN PRI (ITU-T Q.931)	No (Europe only)	Certified	Met all critical CRs and FRs.
	2-Wire Analog Ground Start (GR-506-CORE)	No	Certified	Met all critical CRs and FRs.[11]

NOTES:

1 The SUT T1 CAS interface does not recognize Remove from Service (Busy Out) or Restore to Service (Make Idle) condition from the distant end switch. In addition, when the Busy Out condition is invoked across the T1 CAS interface, it causes the SUT 3845 and 2851gateway T1 CAS interface to deregister from its current subscriber and reregister to an alternate subscriber and then within 1 to 5 minutes repeat the process and go back to its original subscriber. During this transition period, calls are unable to process to the SUT.

2 The SUT does not support NFAS on their ISDN PRI NI2 interface. DISA's adjudication of this discrepancy was completed on 17 December 2008 and was ruled to have a minor operational impact. Furthermore, DISA stated they intent to modify the next update of the UCR to change NFAS for a PBX 1 from required to conditional.

Table 2-4. SUT Interoperability Test Summary (continued)

NOTES (continued):

3 All of the features on the VoIP phones were tested using multiple line appearances. A minimum of two line appearances is required to meet the MLPP interoperability requirements for Call Features with the exception of call hold, call pickup, and call forwarding functions. Call Forward No Answer, Call Forward Busy, and Multi-Line Hunt Service are supported on both VoIP and analog phones. Call Forward Variable, Three-way Calling, Call Hold, Call Pick-up, and Call Transfer are supported on VoIP phones only. These features are required for a PBX 1 for all instruments, however since this requirement is a new UCR requirement and the vendor has 18 months to develop it (July 2010), the operational impact is minor. Since the SUT test window started before the 18 month development window expired, DISA stated this new feature requirement does not apply. All of the features on the VoIP phones were tested using multiple line appearances. A minimum of two line appearances is required to meet the MLPP interoperability requirements for Call Features with the exception of call hold, call pickup, and call forwarding functions. Although the SUT does not support Precedence Call Waiting, they do support multiple call appearances on their VoIP phones. This provides the ability for a user to receive additional calls while active with another call. A short "ping" ring is not provided when calls are forwarded; however, the phone does visually display that call forward variable is enabled. There is a minor operational impact. When a ROUTINE call is placed to a hunt group, and a ring-no-answer condition occurs, the calling party is diverted to the MLPP alternate directory number. This configuration must be done to allow correct treatment to be provided to precedence calls above ROUTINE that are placed to the hunt group. There is no operational impact.

4 The SUT met all IPv6 requirements through testing and LoC with the following exception: The Cisco CP-7940G and CP-7960G end instruments did not meet dual stack IPv6 requirements. These end instruments represent legacy end instruments which are IPv4 only; however, the SUT met the minimum requirement for dual stack IPv6 end instruments with the other IP end instruments listed in this table and a dual stack call control agent in accordance with Reference (g).

5 The SUT only supports emergency service 911 public safety features. The following public safety features are not supported and therefore are not covered in this certification: Trace of terminating calls, Outgoing call trace, Tandem call trace, and Trace of a call in progress. There is no operational impact because these public safety features are not required for a PBX 1.

6 Meet-Me Conferencing can be met through the use of an optional adjunct conferencing system called the Cisco Meeting Place Express which is covered under a separate certification.

7 The SUT does not support an MLPP global diversion number. Each station must be individually configured with a precedence diversion number from a single location using the Bulk Administration Tool provided with the Cisco Unified Communication Manager. The operational impact is minor because diversion settings can be configured for all of the stations provisioned on the switch from a single location.

8 The SUT does not support the Loss of C2 announcement. This announcement is invoked only when a DSN subscriber is automatically routed to a non-MLPP network. DISA adjudicated this anomaly as having a minor operational impact because this announcement would rarely be invoked on a PBX 1. Furthermore, DISA, in coordination with the Joint Staff, stated their intent to modify the next update of the UCR to change the Loss of C2 announcement from required to conditional for a PBX 1.

9 Security is tested by DISA-led Information Assurance test teams and published in a separate report, Reference (c).

10 The following discrepancies noted with the SUT were adjudicated by DISA on 2 September 2009 as having a minor operational impact:

a. The VoIP SUT session control components and end instruments can only have the signaling service Traffic Class configured for 21 different DSCP values and not the full range required of 0-63.
b. The MCS7835 and the MCS7825 call managers OAM traffic is tagged at zero and is not configurable.
c. The 2851 and 3845 gateways are tagging IPv4 RTCP traffic at zero and it is not configurable.
d. When the CP-7940G and CP-7960G phones are powered up, some of the UDP/TFTP traffic has a DSCP value of 4 and 802.1Q value of 5 and can not be changed.
e. The SUT management workstation provided during testing did not assign DSCP values for OAM IP traffic.
f. IP phones are incorrectly tagging IPv6 TCP traffic during power up.
g. Soft Client is incorrectly tagging all traffic during power up.
h. The 802.1Q COS tag values are not independently configurable from the DSCP values.
i. The MCS7825H4 Communication Manager server stopped transmitting IP Traffic. The NIC failover must be disabled to correct this problem. The NIC failover is offered on this server but is not required for a PBX1. NIC failover is not certified for any server platform and should not be enabled. This setting will be annotated in the deployment guide for this server.
j. End Instruments, except for the Telecore 2151, do not support the manual configuration of the IPv6 default gateway.
k. Communication Managers are incorrectly tagging UDP/TFTP traffic to the end instrument after end instrument power up.

11 This interface requirement was met by the vendor's letter of compliance.

Table 2-4. SUT Interoperability Test Summary (continued)

LEGEND:

802.1Q	Standards for Local and Metropolitan Area Networks: Virtual Bridged Local Area Networks	LoC	Letters of Compliance
		LSSGR	Local Access and Transport Area (LATA) Switching Systems Generic Requirements
802.3u	Standard for carrier sense multiple access with collision detection at 100 Mbps	Mbps	Megabits per second
		MCS	Media Convergence Servers
ANSI	American National Standards Institute	MFR1	Multi-Frequency Recommendation 1
APL	Approved Products List	MLPP	Multi-Level Precedence and Preemption
ASLAN	Assured Services Local Area Network	ms	milliseconds
BRI	Basic Rate Interface	NI 1/2	National ISDN Standard 1 or 2
C2	Command and Control	NI2	National ISDN Standard 2
CAS	Channel Associated Signaling	NIC	Network Interface Card
CoS	Class of Service	NFAS	Non Facility Associated Signaling
CP	Cisco Phone	OAM	Operational Administration and Maintenance
CRs	Capability Requirements	PBX 1	Private Branch Exchange 1
DISA	Defense Information Systems Agency	PMO	Program Management Office
DP	Dial Pulse	PNT	Preemption Notification Tone
DSCP	Differentiated Services Code Point	PRI	Primary Rate Interface
DSN	Defense Switched Network	PSTN	Public Switched Telephone Network
DSS1	Digital Subscriber Signaling 1	Q.931	Signaling Standard for ISDN
DTMF	Dual Tone Multi-Frequency	Q.955.3	ISDN Signaling standard for E1 MLPP
E1	European Basic Multiplex Rate (2.048 Mbps)	RTCP	RTP Control Protocol
EI	End Instrument	RTP	Real-time Transport Protocol
FRs	Feature Requirements	SS7	Signaling System 7
GR	Generic Requirement	SUT	System Under Test
GR-506-CORE	LSSGR: Signaling for Analog Interfaces	T1	Digital Transmission Link Level 1 (1.544 Mbps)
ICA	Isolated Code Announcement	T1.607	ISDN Layer 3 Signaling Specification for Circuit Switched Bearer Service for DSS1
IEEE	Institute of Electrical and Electronics Engineers	T1.619a	SS7 and ISDN MLPP Signaling Standard for T1
IP	Internet Protocol	TCP	Transmission Control Protocol
IPv4	Internet Protocol version 4	TFTP	Trivial File Transfer Protocol
IPv6	Internet Protocol version 6	UC	Unified Capabilities
ISDN	Integrated Services Digital Network	UCR	Unified Capabilities Requirements
ITU-T	International Telecommunication Union - Telecommunication Standardization Sector	UDP	User Datagram Protocol
		VoIP	Voice over Internet Protocol
JITC	Joint Interoperability Test Command		

12. TEST AND ANALYSIS REPORT. No detailed test report was developed in accordance with the Program Manager's request. JITC distributes interoperability information via the JITC Electronic Report Distribution (ERD) system, which uses Unclassified-But-Sensitive Internet Protocol Router Network (NIPRNet) e-mail. More comprehensive interoperability status information is available via the JITC System Tracking Program (STP). The STP is accessible by .mil/gov users on the NIPRNet at https://stp.fhu.disa.mil. Test reports, lessons learned, and related testing documents and references are on the JITC Joint Interoperability Tool (JIT) at http://jit.fhu.disa.mil (NIPRNet), or http://199.208.204.125 (SIPRNet). Information related to DSN testing is on the Telecom Switched Services Interoperability (TSSI) website at http://jitc.fhu.disa.mil/tssi. Due to the sensitivity of the information, the Information Assurance Accreditation Package (IAAP) that contains the approved configuration and deployment guide must be requested directly through government civilian or uniformed military personnel from the Unified Capabilities Certification Office (UCCO), e-mail: ucco@disa.mil.

Table 2-5. SUT Interoperability Requirements/Status

DSN Trunk Interfaces							
Interface	**Critical**	**Interface Status**	**UCR Requirement**		**Reference**	**Test Results**	**Remarks**
T1 CAS (MFR1, DTMF, DP)	No	Not Certified (See note 1.)	Trunking	Direct Inward Dialing (C)	UCR Section 5.2.1.3.2	Met	
				Trunk Group-Remove from Service (C)	UCR Section 5.2.1.5.5	Not Certified	See note 1.
				Trunk Group-Restore to Service (C)	UCR Section 5.2.1.5.5	Not Certified	See note 1.
				Normal Wink Start Operations (C)	UCR Section 5.2.4.3.3.1.1	Met	
				Glare Operation (C)	UCR Section 5.2.4.3.3.1.2	Met	
				Abnormal Wink Start (C)	UCR Section 5.2.4.3.3.2.1	Met	
				Glare Resolution (C)	UCR Section 5.2.4.3.3.2.2	Met	
				Call for Service Timing (R)	UCR Section 5.2.4.3.5	Met	
				Guard Timing (R)	UCR Section 5.2.4.3.6	Met	
				Satellite Timing (C)	UCR Section 5.2.4.3.7	Met	
				Disconnect Control (C)	UCR Section 5.2.4.3.8	Met	
				Reselect and Retrial (C)	UCR Section 5.2.4.3.9	Not Tested	
				Off-Hook Supervision Transition (C)	UCR Section 5.2.4.3.10	Met	
				Dial-Pulse Signals (C)	UCR Section 5.2.4.4.1	Met	
				DTMF Signaling (C)	UCR Section 5.2.4.4.2	Met	
				Standard Digit Format for Precedence (C)	UCR Section 5.2.4.4.2.1	Met	
				MFR1 2/6 Signaling (C)	UCR Section 5.2.4.4.3	Met	
				Alerting Signals and Tones (R)	UCR Section 5.2.4.5.1	Met	
				DSN Transmission Interface (R)	UCR Section 5.2.5	Met	
				PCM-24 Digital Trunk Interface (R)	UCR Section 5.2.6.1	Met	
				Interface Characteristics (R)	UCR Section 5.2.6.1.1	Met	
				Supervisory Channel Associated Signaling (C)	UCR Section 5.2.6.1.2	Met	
				Clear Channel Capability (R)	UCR Section 5.2.6.1.3	Met	
				Alarm and Restoral Requirements (R)	UCR Section 5.2.6.1.4	Met	
				Interoperation of PCM-24 and PCM-30 (C)	UCR Section 5.2.6.3	Not Tested	See note 2.
				Integrated Digital Loop Carrier (C)	UCR Section 5.2.6.5	Met	
			Voice	MOS (R)	CJCSI 6215.01C	Met	
				Secure calls (R)	CJCSI 6215.01C	Met	
			Facsimile	Analog: ITU-T T.4 (R)	DISR	Met	
			Data	Modem (VBD) (R)	CJCSI 6215.01C	Met	
				Secure data (STE/STU-III) (R)	CJCSI 6215.01C	Met	

Table 2-5. SUT Interoperability Requirements/Status (continued)

DSN Trunk Interfaces							
Interface	**Critical**	**Interface Status**	**UCR Requirement**		**Reference**	**Test Results**	**Remarks**
E1 CAS (MFR1, DTMF, DP)	No (Europe only)	Not Tested (See note 3.)	Trunking	Direct Inward Dialing (C)	UCR Section 5.2.1.3.1	Not Tested	
				Trunk Group-Remove from Service (C)	UCR Section 5.2.1.5.5	Not Tested	
				Trunk Group-Restore to Service (C)	UCR Section 5.2.1.5.5	Not Tested	
				Normal Wink Start Operations (C)	UCR Section 5.2.4.3.3.1.1	Not Tested	
				Glare Operation (C)	UCR Section 5.2.4.3.3.1.2	Not Tested	
				Abnormal Wink Start (C)	UCR Section 5.2.4.3.3.2.1	Not Tested	
				Glare Resolution (C)	UCR Section 5.2.4.3.3.2.2	Not Tested	
				Call for Service Timing (R)	UCR Section 5.2.4.3.5	Not Tested	
				Guard Timing (R)	UCR Section 5.2.4.3.6	Not Tested	
				Satellite Timing (C)	UCR Section 5.2.4.3.7	Not Tested	
				Disconnect Control (C)	UCR Section 5.2.4.3.8	Not Tested	
				Reselect and Retrial (C)	UCR Section 5.2.4.3.9	Not Tested	
				Off-Hook Supervision Transition (C)	UCR Section 5.2.4.3.10	Not Tested	
				Dial-Pulse Signals (C)	UCR Section 5.2.4.4.1	Not Tested	
				DTMF Signaling (C)	UCR Section 5.2.4.4.2	Not Tested	
				Standard Digit Format for Precedence (C)	UCR Section 5.2.4.4.2.1	Not Tested	
				MFR1 2/6 Signaling (C)	UCR Section 5.2.4.4.3	Not Tested	
				Alerting Signals and Tones (R)	UCR Section 5.2.4.5.1	Not Tested	
				DSN Transmission Interface (R)	UCR Section 5.2.5	Met	
				PCM-30 Digital Trunk Interface (C)	UCR Section 5.2.6.2	Not Tested	
				Interoperation of PCM-24 and PCM-30 (C)	UCR Section 5.2.6.3	Not Tested	
				Integrated Digital Loop Carrier (C)	UCR Section 5.2.6.5	Not Tested	
			Voice	MOS (R)	CJCSI 6215.01C	Not Tested	
				Secure calls (R)	CJCSI 6215.01C	Not Tested	
			Facsimile	Analog: ITU-T T.4 (R)	DISR	Not Tested	
			Data	Modem (VBD) (R)	CJCSI 6215.01C	Not Tested	
				Secure data (STE/STU-III) (R)	CJCSI 6215.01C	Not Tested	

Table 2-5. SUT Interoperability Requirements/Status (continued)

DSN Trunk Interfaces							
Interface	**Critical**	**Interface Status**	**UCR Requirement**		**Reference**	**Test Results**	**Remarks**
T1 ISDN PRI NI 1/2 (ANSI T1.619a)	Yes	Certified	Trunking	Direct Inward Dialing (C)	UCR Section 5.2.1.3.2	Met	
				National ISDN 1/2 Primary Access (R)	UCR Section 5.2.1.3.4.1	Met	See note 4.
				ISDN ANSI MLPP Service Capability (R)	UCR Section 5.2.1.3.4.1.1	Met	
				Trunk Group-Remove from Service (C)	UCR Section 5.2.1.5.5	Met	
				Trunk Group-Restore to Service (C)	UCR Section 5.2.1.5.5	Met	
				Call for Service Timing (R)	UCR Section 5.2.4.3.5	Met	
				Alerting Signals and Tones (R)	UCR Section 5.2.4.5.1	Met	
				DSN ISDN User-to-Network Signaling (R)	UCR Section 5.2.4.7.1.4.2	Met	
				Application (R)	UCR Section 5.2.4.7.1.1	Met	
				Physical Layer (R)	UCR Section 5.2.4.7.1.2	Met	
				Data Link Layer (R)	UCR Section 5.2.4.7.1.3	Met	
				Data Link Connection (R)	UCR Section 5.2.4.7.1.3.1	Met	
				Peer-to-Peer Procedures of Data-Link Layer (R)	UCR Section 5.2.4.7.1.3.2	Met	
				Layer 3 DSN User-to-Network Signaling (R)	UCR Section 5.2.4.7.1.4	Met	
				DSN User-to-Network Signaling for Circuit-Switched Bearer Services (R)	UCR Section 5.2.4.7.1.4.2	Met	
				Sequence of Messages for DSN Circuit-Switched Calls (R)	UCR Section 5.2.4.7.1.4.3	Met	
				Message Functional Definition and Content (R)	UCR Section 5.2.4.7.1.4.4	Met	
				General Message Format and Information Elements Coding (R)	UCR Section 5.2.4.7.1.4.5	Met	
				Supplementary Services (C)	UCR Section 5.2.4.7.1.4.6	Not Tested	See note 2.
				DSN Transmission Interface (R)	UCR Section 5.2.5	Met	
				PCM-24 Digital Trunk Interface (R)	UCR Section 5.2.6.1	Met	
				Interface Characteristics (R)	UCR Section 5.2.6.1.1	Met	
				Clear Channel Capability (R)	UCR Section 5.2.6.1.3	Met	
				Alarm and Restoral Requirements (R)	UCR Section 5.2.6.1.4	Met	
				Interoperation of PCM-24 and PCM-30 (C)	UCR Section 5.2.6.3	Met	
				Integrated Digital Loop Carrier (C)	UCR Section 5.2.6.5	Met	
			Voice	MOS (R)	CJCSI 6215.01C	Met	
				Secure calls (R)	CJCSI 6215.01C	Met	
			Facsimile	Analog: ITU-T T.4 (R)	DISR	Met	

Table 2-5. SUT Interoperability Requirements/Status (continued)

DSN Trunk Interfaces							
Interface	**Critical**	**Interface Status**	**UCR Requirement**		**Reference**	**Test Results**	**Remarks**
T1 ISDN PRI NI 1/2 (ANSI T1.619a) (continued)	Yes	Certified	Data	Modem (VBD) (R)	CJCSI 6215.01C	Met	
				56 kbps switched data (R: PRI only)	UCR Section 5.2.2.9.6	Not Tested	
				64 kbps switched data (R: PRI only)	UCR Section 5.2.2.9.6	Not Tested	
				NX56 synchronous BER (R: PRI only)	UCR Section 5.2.2.9.6	Not Tested	
				NX64 synchronous BER (R: PRI only)	UCR Section 5.2.2.9.6	Not Tested	
				Secure data (STE/STU-III) (R)	CJCSI 6215.01C	Met	
			VTC	ITU-T H.320 (R: PRI only)	FTR 1080B-2002	Not Tested	See note 2.

Table 2-5. SUT Interoperability Requirements/Status (continued)

DSN Trunk Interfaces							
Interface	**Critical**	**Interface Status**	**UCR Requirement**		**Reference**	**Test Results**	**Remarks**
E1 ISDN PRI (ITU-T Q.955.3)	No (Europe only)	Not certified (See note 5.)	Trunking	Direct Inward Dialing (C)	UCR Section 5.2.1.3.2	Met	
				ITU-T ISDN Primary Access (C)	UCR Section 5.2.1.3.4.2	Met	
				ITU-T ISDN Primary Access Digital Subscriber Signaling System Number 1 MLPP (C)	UCR Section 5.2.1.3.4.2.1	Not Tested	See note 5.
				Trunk Group-Remove from Service (C)	UCR Section 5.2.1.5.5	Not Tested	See note 2.
				Trunk Group-Restore to Service (C)	UCR Section 5.2.1.5.5	Not Tested	See note 2.
				Call for Service Timing (R)	UCR Section 5.2.4.3.5	Met	
				Disconnect Control (C)	UCR Section 5.2.3.4.8	Met	
				Off-Hook Supervision Transition (C)	UCR Section 5.2.3.4.10	Met	
				DSN ISDN User-to-Network Signaling (R)	UCR Section 5.2.4.7.1.4.2	Met	
				Application (R)	UCR Section 5.2.4.7.1.1	Met	
				Physical Layer (R)	UCR Section 5.2.4.7.1.2	Met	
				Data Link Layer (R)	UCR Section 5.2.4.7.1.3	Met	
				Data Link Connection (R)	UCR Section 5.2.4.7.1.3.1	Met	
				Peer-to-Peer Procedures of Data-Link Layer (R)	UCR Section 5.2.4.7.1.3.2	Met	
				Layer 3 DSN User-to-Network Signaling (R)	UCR Section 5.2.4.7.1.4	Met	
				DSN User-to-Network Signaling for Circuit-Switched Bearer Services (R)	UCR Section 5.2.4.7.1.4.2	Met	
				Sequence of Messages for DSN Circuit-Switched Calls (R)	UCR Section 5.2.4.7.1.4.3	Met	
				Message Functional Definition and Content (R)	UCR Section 5.2.4.7.1.4.4	Met	
				General Message Format and Information Elements Coding (R)	UCR Section 5.2.4.7.1.4.5	Met	
				PCM-30 Digital Trunk Interface (C)	UCR Section 5.2.6.2	Met	
				Interoperation of PCM-24 and PCM-30 (C)	UCR Section 5.2.6.3	Not Tested	See note 2.
				Integrated Digital Loop Carrier (C)	UCR Section 5.2.6.5	Not Tested	See note 2.
			Voice	MOS (R)	CJCSI 6215.01C	Met	
				Secure calls (R)	CJCSI 6215.01C	Met	

Table 2-5. SUT Interoperability Requirements/Status (continued)

DSN Trunk Interfaces							
Interface	**Critical**	**Interface Status**	**UCR Requirement**		**Reference**	**Test Results**	**Remarks**
E1 ISDN PRI (ITU-T Q.955.3) (continued)	No (Europe only)	Not Certified (See note 5.)	Facsimile	Analog: ITU-T T.4 (R)	DISR	Met	
			Data	Modem (VBD) (R)	CJCSI 6215.01C	Met	
				56 kbps switched data (R: PRI only)	UCR Section 5.2.2.9.6	Not Tested	See note 2.
				64 kbps switched data (R: PRI only)	UCR Section 5.2.2.9.6	Not Tested	See note 2.
				NX56 synchronous BER (R: PRI only)	UCR Section 5.2.2.9.6	Not Tested	See note 2.
				NX64 synchronous BER (R: PRI only)	UCR Section 5.2.2.9.6	Not Tested	See note 2.
				Secure data (STE/STU-III) (R)	CJCSI 6215.01C	Met	
			VTC	ITU-T H.320 (R: PRI only)	FTR 1080B-2002	Not Tested	See note 2.

Table 2-5. SUT Interoperability Requirements/Status (continued)

DSN Line Interfaces							
Interface	Critical	Interface Status	UCR Requirement		Reference	Test Results	Remarks
2-Wire Loop Start Analog	Yes	Certified	Access	Directory Number Identification (R)	UCR Section 5.2.1.1.1	Met	
				PBX Line (C)	UCR Section 5.2.1.3.1	Met	
				Analog Line (R)	UCR Section 5.2.1.3.5	Met	
				Basic Line Test Capabilities (R)	UCR Section 5.2.1.5.4.1.1	Met	
				Advanced Line Test Capabilities (C)	UCR Section 5.2.1.5.4.1.1	Not Tested	See note 2.
				Loop Start Line (R: 2-Wire Analog only)	UCR Section 5.2.4.2.1	Met	
				Reverse Battery (R)	UCR Section 5.2.4.3.1	Met	
				Alerting Signals and Tones (R)	UCR Section 5.2.4.5.1	Met	
			Voice	MOS (R)	CJCSI 6215.01C	Met	
				Secure calls (R)	CJCSI 6215.01C	Met	
			Facsimile	Analog: ITU-T T.4 (R)	DISR	Met	
			Data	Modem (VBD) (R)	CJCSI 6215.01C	Met	
				Secure data (STE/STU-III) (R)	CJCSI 6215.01C	Met	
ISDN BRI NI 1/2 (ANSI T1.619a)	No	Not Tested (See note 6.)	Access	Directory Number Identification (R)	UCR Section 5.2.1.1.1	Not Tested	
				National ISDN 1/2 Basic Access (C)	UCR Section 5.2.1.3.3	Not Tested	
				Alerting Signals and Tones (R)	UCR Section 5.2.4.5.1	Not Tested	
				S/T Reference Point (R)	UCR Section 5.2.4.7.1.2.1	Not Tested	
			Voice	MOS (R)	CJCSI 6215.01C	Not Tested	
				Secure calls (R)	CJCSI 6215.01C	Not Tested	
			Facsimile	Analog: ITU-T T.4 (R)	DISR	Not Tested	
			Data	Modem (VBD) (R)	CJCSI 6215.01C	Not Tested	
				Secure data (STE/STU-III) (R)	CJCSI 6215.01C	Not Tested	
			VTC	ITU-T H.320 (R: BRI only)	FTR 1080B-2002	Not Tested	
2-Wire Proprietary Digital	No	Not Tested (See note 6.)	Access	Directory Number Identification (R)	UCR Section 5.2.1.1.1	Not Tested	
				Alerting Signals and Tones (R)	UCR Section 5.2.4.5.1	Not Tested	
			Voice	MOS (R)	CJCSI 6215.01C	Not Tested	
				Secure calls (R)	CJCSI 6215.01C	Not Tested	

Table 2-5. SUT Interoperability Requirements/Status (continued)

DSN Features and Capabilities						
Feature/ Capability	**Critical**	**Feature Status**	**UCR Requirement**	**Reference**	**Test Results**	**Remarks**
Common Features	Yes	Certified	Individual Lines (R)	UCR Section 5.2.1.1.1	Met	
			Denied originating service (C)	UCR Section 5.2.1.1.3	Not Tested	See note 2.
			Code restriction and diversion (C)	UCR Section 5.2.1.1.4	Met	
			Call waiting (R)	UCR Section 5.2.1.1.5.1	Met	See note 7.
			Three-way calling (R)	UCR Section 5.2.1.1.6	Met	See note 7.
			Add-on transfer, conference calling, and call hold (C)	UCR Section 5.2.1.1.7	Met	See note 7.
			Call Transfer Individual – All calls (R)	UCR Section 5.2.1.1.7.1	Met	See note 7.
			Call Transfer - Internal Only (R)	UCR Section 5.2.1.1.7.2	Met	See note 7.
			Call Transfer – Individual – Incoming Only/Add-On Consultation Hold – Incoming Call (R)	UCR Section 5.2.1.1.7.3	Met	See note 7.
			Call Transfer – Outside (R)	UCR Section 5.2.1.1.7.4	Met	See note 7.
			Call Transfer – Add-On Restricted Station (C)	UCR Section 5.2.1.1.7.5	Not Tested	See note 2.
			Call Transfer – Attendant (C)	UCR Section 5.2.1.1.7.6	Not Tested	See note 2.
			Call Hold (R)	UCR Section 5.2.1.1.7.7	Met	See note 7.
			Conference Calling – Six Way Station Controlled (C)	UCR Section 5.2.1.1.7.8	Met	
			Call Forwarding Variable (R)	UCR Section 5.2.1.1.8.1	Met	See note 7.
			Call Forward Busy Line (R)	UCR Section 5.2.1.1.8.2	Met	See note 7.
			Call Forwarding – Don't Answer – All Calls (R)	UCR Section 5.2.1.1.8.3	Met	See note 7.
			Selective Call Forwarding (C)	UCR Section 5.2.1.1.8.4	Met	
			Call pick-up (C)	UCR Section 5.2.1.1.9.1	Met	See note 7.
			Address Translation (C)	UCR Section 5.2.1.7	Met	
			Assured Dial Tone (C)	UCR Section 5.2.1.9	Met	
Attendant	No	Not Tested	Attendant Features (C)	UCR Section 5.2.1.2.2	Not Tested	See note 2.
Public Safety	Yes	Certified	Emergency Service (911) Caller (R)	UCR Section 5.2.1.4.1.1	Met	See note 8.
			Emergency Service (911) Public Safety Answering Service (C)	UCR Section 5.2.1.4.1.2	Not Tested	See note 8.
			Enhanced Emergency Service (E911) (C)	UCR Section 5.2.1.4.1.3	Not Tested	See note 8.
			Trace of terminating calls (C)	UCR Section 5.2.1.4.2	Not Tested	See note 8.
			Outgoing call trace (C)	UCR Section 5.2.1.4.3	Not Tested	See note 8.

Table 2-5. SUT Interoperability Requirements/Status (continued)

DSN Features and Capabilities						
Feature/ Capability	**Critical**	**Feature Status**	**UCR Requirement**	**Reference**	**Test Results**	**Remarks**
Conferencing	No	Not Certified	Preset Conferencing (C)	UCR Section 5.2.1.6	Not Tested	See note 2.
			Meet-Me Conferencing (R)	UCR Section 5.2.1.6.2	Met	See note 9.
			Progressive Conferencing (C)	UCR Section 5.2.1.6.3	Not Tested	See note 2.
Nailed-up Connections	No	Not Tested	Nailed-Up Connections (C)	UCR Section 5.2.1.8	Not Tested	See note 2.
DSN Hotline Services	No	Certified	DSN Analog Hotline Service (C)	UCR Section 5.2.1.12	Not Tested	See note 2.
MLPP	Yes	Certified	MLPP Overview (R)	UCR Section 5.2.2.1.1	Met	
			Preemption in the Network (R)	UCR Section 5.2.2.2	Met	
			Network Facility with Lower Precedence Calls (R)	UCR Section 5.2.2.2.1	Met	
			Network Facility with Equal or Higher Precedence Calls (R)	UCR Section 5.2.2.2.2	Met	
			Precedence Call Diversion (R)	UCR Section 5.2.2.3	Met	See note 10.
			Channel Associated Signaling (C)	UCR Section 5.2.2.4.1	Met	See note 1.
			Primary Rate Interface (R)	UCR Section 5.2.2.4.2	Met	
			Analog Line MLPP (R)	UCR Section 5.2.2.5	Met	
			ISDN MLPP Basic Rate Interface (C)	UCR Section 5.2.2.6	Not Tested	See note 6.
			ISDN Primary Rate Interface (R)	UCR Section 5.2.2.7	Met	
			Precedence Call Waiting (R)	UCR Section 5.2.2.8.1	Met	See note 7.
			Call Forwarding (R)	UCR Section 5.2.2.8.2	Met	See note 7.
			Call Transfer (R)	UCR Section 5.2.2.8.3	Met	See note 7.
			Call Hold (R)	UCR Section 5.2.2.8.4	Met	See note 7.
			Three-Way Calling (R)	UCR Section 5.2.2.8.5	Met	See note 7.
			Call Pickup (C)	UCR Section 5.2.2.8.6	Met	See note 7.
			Conferencing (C)	UCR Section 5.2.2.8.7.1	Met	See note 9.
			Multiline Hunt Group (C)	UCR Section 5.2.2.8.8	Met	
			Community of Interest (C)	UCR Section 5.2.2.8.9	Not Tested	See note 2.
			MLPP Interaction with EKTS features (C)	UCR Section 5.2.2.10.1	Not Tested	See note 2.

Table 2-5. SUT Interoperability Requirements/Status (continued)

DSN Features and Capabilities						
Feature/ Capability	**Critical**	**Feature Status**	**UCR Requirement**	**Reference**	**Test Results**	**Remarks**
Call Processing	Yes	Certified	Call Treatments (R)	UCR Section 5.2.3.1	Met	
			Primary and Alternate Routing (C)	UCR Section 5.2.3.2	Met	
			E&M Lead Signaling States (C)	UCR Section 5.2.3.3.1	Not Tested	See note 6.
			4-Wire Analog User Access Lines (C)	UCR Section 5.2.3.3.2	Not Tested	See note 6.
			2-Wire User Access Lines (R)	UCR Section 5.2.3.3.3	Met	
			Termination of Analog Lines (R)	UCR Section 5.2.3.3.4	Met	
			DSN User Dialing (R)	UCR Section 5.2.3.5.1.1	Met	
			Interswitch and Intraswitch Dialing (R)	UCR Section 5.2.3.5.1.1	Met	
			Seven-Digit Dialing (R)	UCR Section 5.3.3.5.2.1	Met	
			Ten-Digit Dialing (R)	UCR Section 5.2.3.5.2.2	Met	
			Access Code (R)	UCR Section 5.2.3.5.1.3	Met	
			Access Digit (R)	UCR Section 5.2.3.5.1.3.1	Met	
			Precedence Digit (R)	UCR Section 5.2.3.5.1.3.2	Met	
			Service Digit (R)	UCR Section 5.2.3.5.1.3.3	Met	
			Route Code (R)	UCR Section 5.2.3.5.1.4	Met	
			Area Code (R)	UCR Section 5.2.3.5.1.5	Met	
			Switch Code (R)	UCR Section 5.2.3.5.1.6	Met	
			Line Number (R)	UCR Section 5.2.3.5.1.7	Met	
			Calling Name Delivery (C)	UCR Section 5.2.3.5.1.8.1	Not Tested	See note 2.
			Calling Number Delivery (R)	UCR Section 5.2.3.5.1.8.2	Met	
			Emergency Service 911 Conflict Resolution (R)	UCR Section 5.2.3.5.1.9	Met	
			DSN Switch Outpulsing Digit Formats (C)	UCR Section 5.2.3.5.2	Met	See note 1.
			Standard Directory Number (R)	UCR Section 5.2.3.5.3	Met	
			Standard Test Numbers (C)	UCR Section 5.2.3.5.4	Not Tested	See note 2.
			Base Services – Abbreviated Numbers (C)	UCR Section 5.2.3.5.5	Not Tested	See note 2.
			Digit Reception Requirements (R)	UCR Section 5.2.3.5.6	Met	
			Screening (C)	UCR Section 5.2.3.5.8	Met	
ISDN Services	Yes	Certified	BRI Access, Call Control and Signaling (C)	UCR Section 5.2.9.2, Table 5.2.9-1	Not Tested	See note 6.
			Uniform Interface Configuration for BRIs (C)	UCR Section 5.2.9.2, Table 5.2.9-2	Not Tested	See note 6.
			Electronic Key Telephone Systems (EKTS) (C)	UCR Section 5.2.9.2, Table 5.2.9-3	Not Tested	See note 6.
			PRI Access, Call Control and Signaling (R)	UCR Section 5.2.9.2, Table 5.2.9-4	Met	See note 4.
			PRI Features (R)	UCR Section 5.2.9.2, Table 5.2.9-5	Met	See note 4.
			Packet Data Features and Capabilities (C)	UCR Section 5.2.9.2, Table 5.2.9-6	Not Tested	See note 6.

Table 2-5. SUT Interoperability Requirements/Status (continued)

DSN Features and Capabilities						
Feature/ Capability	**Critical**	**Feature Status**	**UCR Requirement**	**Reference**	**Test Results**	**Remarks**
Synchronization	Yes	Certified	Line timing mode (R)	UCR Section 5.2.11.2	Met	
			Internal Stratum 4 (R)	UCR Section 5.2.10.1.1.2.2	Met	
			Synchronization Performance Monitoring Criteria (C)	UCR Section 5.2.10.2	Not Tested	See note 2.
			DS1 Traffic Interfaces (C)	UCR Section 5.2.10.3	Not Tested	See note 2.
			DS0 Traffic Interconnects (C)	UCR Section 5.2.10.4	Not Tested	See note 2.
Reliability	Yes	Certified	System Availability (R)	UCR Section 5.2.11.2	Met	
			Backup Power (R)	UCR Section 5.2.11.3	Not Tested	See note 11.
			Power Components (R)	UCR Section 5.2.11.3.1	Not Tested	See note 11.
			UPS Requirements (R)	UCR Section 5.2.11.3.2	Not Tested	See note 11.
			UPS PBX 1 Load Capacity (R)	UCR Section 5.2.11.3.2.1	Not Tested	See note 11.
			Backup Power (Environmental) (R)	UCR Section 5.2.11.3.3	Not Tested	See note 11.
			Alarms (R)	UCR Section 5.2.11.3.4	Not Tested	See note 11.
Security	Yes	Certified	GR-815, STIGs, and DoDI 8510.bb (DIACAP) (R)	UCR Section 3	Met	See note 12.
VoIP						
Feature/ Capability	**Critical**	**Feature Status**	**UCR Requirement**	**Reference**	**Test Results**	**Remarks**
VoIP System	No	Certified (See note 13.)	Voice Quality with MOS of 4.0 or better (R)	UCR Section 5.2.12.8.2.1	Met	
			ITU-T G.711 PCM CODEC (R)	UCR Section 5.2.12.8.2.2	Met	
			MLPP (R)	UCR Section 5.2.12.8.2.3	Met	
			Security (R)	UCR Section 5.2.12.8.2.4	Met	
			Network management (C)	UCR Section 5.2.12.8.2.5	Met	
			System timing (R)	UCR Section 5.2.12.8.2.6	Met	
			Latency ≤ 60 milliseconds (R)	UCR Section 5.2.12.8.2.7	Met	
			IPv6 capable (R)	UCR Section 5.2.12.8.2.8	Met	See notes 14 and 15 f, j, k.
			Service Class Tagging (R)	UCR Section 5.2.12.8.2.9	Met	See notes 15 a, b, c, d, e, f, g, h, k, l.
			VoIP System Downtime (IP network 80 min/yr Subscriber 20 min/yr) (R)	UCR Section 5.2.12.8.2.10	Met	See note 15 i.
			Softphone Requirements (R)	DISA Memo (Reference h)	Met	See note 16.

Table 2-5. SUT Interoperability Requirements/Status (continued)

Network Gateways							
Interface	**Critical**	**Interface Status**	**UCR Requirement**		**Reference**	**Test Results**	**Remarks**
PSTN (See note 17.)	No	Certified	Trunking	Positive Identification Control (C)	CJCSI 6215.01C	Met	
				On-Netting (C)	CJCSI 6215.01C	Met	
				Off-Netting (C)	CJCSI 6215.01C	Met	
				Ground Start Line (R)	UCR Section 5.2.2	Met	See note 18.
				Immediate Start (C)	UCR Section 5.3.2	Met	
				Delay Dial (C)	UCR Section 5.3.4	Met	

NOTES:

1 The SUT T1 CAS interface does not recognize Remove from Service (Busy Out) or Restore to Service (Make Idle) condition from the distant end switch. In addition, when the Busy Out condition is invoked across the T1 CAS interface, it causes the SUT 3845 and 2851gateway T1 CAS interface to deregister from its current subscriber and reregister to an alternate subscriber and then within 1 to 5 minutes repeat the process and go back to its original subscriber. During this transition period, calls are unable to process to the SUT.

2 This feature/capability is not supported by the SUT. This is not a required feature for a PBX 1. There is no risk associated with the SUT not supporting this feature.

3 This interface is supported by the SUT; however, it was not tested. This interface is therefore not certified by JITC. This is not a required interface for a PBX 1.

4 The SUT does not support NFAS on their ISDN PRI NI2 interface. DISA's adjudication of this discrepancy was completed on 17 December 2008 and was ruled to have a minor operational impact. Furthermore, DISA stated they intent to modify the next update of the UCR to change NFAS for a PBX 1 from required to conditional.

5 The E1 ISDN PRI interface is supported by the SUT; however, it does not support ITU-T Q.955.3 MLPP. The SUT E1 ISDN PRI interface is therefore not certified by JITC for use within the DSN. This interface is certified only for PSTN. This is not a required DSN interface for a PBX 1.

6 This interface is not supported by the SUT. This is not a required interface for a PBX 1. There is no risk associated with the SUT not supporting this interface.

7 All of the features on the VoIP phones were tested using multiple line appearances. A minimum of two line appearances is required to meet the MLPP interoperability requirements for Call Features with the exception of call hold, call pickup, and call forwarding functions. Call Forward No Answer, Call Forward Busy, and Multi-Line Hunt Service are supported on both VoIP and analog phones. Call Forward Variable, Three-way Calling, Call Hold, Call Pick-up, and Call Transfer are supported on VoIP phones only. These features are required for a PBX 1 for all instruments, however since this requirement is a new UCR requirement and the vendor has 18 months to develop it (July 2010), the operational impact is minor. Since the SUT test window started before the 18 month development window expired, DISA stated this new feature requirement does not apply. All of the features on the VoIP phones were tested using multiple line appearances. Although the SUT does not support Precedence Call Waiting, they do support multiple call appearances on their VoIP phones. This provides the ability for a user to receive additional calls while active with another call. A short "ping" ring is not provided when calls are forwarded; however, the phone does visually display that call forward variable is enabled. There is a minor operational impact. When a ROUTINE call is placed to a hunt group, and a ring-no-answer condition occurs, the calling party is diverted to the MLPP alternate directory number. This configuration must be done to allow correct treatment to be provided to precedence calls above ROUTINE that are placed to the hunt group. There is no operational impact.

8 The SUT only supports emergency service 911 public safety features. The following public safety features are not supported and therefore are not covered in this certification: Trace of terminating calls, Outgoing call trace, Tandem call trace, and Trace of a call in progress. There is no operational impact because these public safety features are not required for a PBX 1.

9 Meet-Me Conferencing can be met through the use of an optional adjunct conferencing system called the Cisco Meeting Place Express which is covered under a separate certification.

10 The SUT does not support an MLPP global diversion number. Each station must be individually configured with a precedence diversion number from a single location using the Bulk Administration Tool provided with the Cisco Unified Communication Manager. The operational impact is minor because diversion settings can be configured for all of the stations provisioned on the switch from a single location.

11 This requirement is a non-testable requirement. It is the responsibility of the respective base/post/camp/station communications agency to provide this with the SUT when installed.

12 Security is tested by DISA-led Information Assurance test teams and published in a separate report, Reference (c).

13 The SUT is certified for VoIP specifically with any certified ASLAN or ASLAN components posted on the UC APL.

14 The SUT met all IPv6 requirements through testing and Letters of Compliance with the following exception: The Cisco CP-7940G and CP-7960G end instruments did not meet dual stack IPv6 requirements. These end instruments represent legacy end instruments which are IPv4 only; however, the SUT met the minimum requirement for dual stack IPv6 end instruments with the other IP end instruments listed in this table and a dual stack call control agent in accordance with Reference (g).

Table 2-5. SUT Interoperability Requirements/Status (continued)

NOTES (continued):

15 The following discrepancies noted with the SUT were adjudicated by DISA on 2 September 2009 as having a minor operational impact:
- a. The VoIP SUT session control components and end instruments can only have the signaling service Traffic Class configured for 21 different DSCP values and not the full range required of 0-63.
- b. The MCS7835 and the MCS7825 call managers OAM traffic is tagged at zero and is not configurable.
- c. The 2851 and 3845 gateways are tagging IPv4 RTCP traffic at zero and it is not configurable.
- d. When the CP-7940G and CP-7960G phones are powered up, some of the UDP/TFTP traffic has a DSCP value of 4 and 802.1Q value of 5 and can not be changed.
- e. The SUT management workstation provided during testing did not assign DSCP values for OAM IP traffic.
- f. IP phones are incorrectly tagging IPv6 TCP traffic during power up.
- g. Soft Client is incorrectly tagging all traffic during power up.
- h. The 802.1Q COS tag values are not independently configurable from the DSCP values.
- i. The MCS7825H4 Communication Manager server stopped transmitting IP Traffic. The NIC failover must be disabled to correct this problem. The NIC failover is offered on this server but is not required for a PBX 1. NIC failover is not certified for any server platform and should not be enabled. This setting will be annotated in the deployment guide for this server.
- j. End Instruments, except for the Telecore 2151, do not support the manual configuration of the IPv6 default gateway.
- k. The 2851 and 3845 gateways cannot set the IPv6 flow label value to zero for RTP media traffic.
- l. Communication Managers are incorrectly tagging UDP/TFTP traffic to the end instrument after end instrument power up.

16 Reference (h) is a DISA memo that stipulates interim softphone requirements that supersede the current UCR 2008 requirements until they are implemented in Change 1. The softphone shall be functionally identical to a traditional IP "Hard" telephone and will be required to provide voice features and functionality provided by a traditional IP "Hard" Telephone with following exceptions:
- a. Audible and visual alerting to the end user of an incoming call, even if the application is running in the background.
- b. Softphone application shall be exempt from reliability, availability and performance (packet loss, jitter, latency) requirements.
- c. Microphone and speaker or headphone, or any other audio input/output device, Ethernet interface(s), and mouse (point and click) interaction.
- d. IPv6 is not required.

17 Voice, facsimile, data, and VTC service requirements for PSTN are identical to DSN with the exception of MLPP.

18 This interface requirement was met by the vendor's letter of compliance.

Table 2-5. SUT Interoperability Requirements/Status (continued)

LEGEND:

ANSI	American National Standards Institute
APL	Approved Products List
ASLAN	Assured Services Local Area Network
BER	Bit Error Ratio
BRI	Basic Rate Interface
C	Conditional
CAS	Channel Associated Signaling
CJCSI	Chairman of the Joint Chiefs of Staff Instruction
CODEC	Coder/Decoder
CP	Cisco Phone
DIACAP	DoD Information Assurance Certification and Accreditation Process
DISA	Defense Information Systems Agency
DISR	DoD IT Standards Registry
DoD	Department of Defense
DoDI	Department of Defense Instruction
DP	Dial Pulse
DS0	Digital Signal Level 0 (64 kbps)
DS1	Digital Signal Level 1 (1.544 Mbps) (2.048 Mbps European)
DSCP	Differentiated Services Code Point
DSN	Defense Switched Network
DTMF	Dual Tone Multi-Frequency
E&M	Ear and Mouth
E1	European Basic Multiplex Rate (2.048 Mbps)
EI	End Instrument
EKTS	Electronic Key Telephone System
FTR	Federal Telecommunications Recommendation
FTR 1080B-2002	Video Teleconferencing Services
G.711	PCM of voice frequencies
GR	Generic Requirement
GR-815	Generic Requirements For Network Element/Network System (NE/NS) Security
H.320	Standard for Narrowband VTC
IP	Internet Protocol
IPv4	Internet Protocol version 4
IPv6	Internet Protocol version 6
ISDN	Integrated Services Digital Network
IT	Information Technology
ITU-T	International Telecommunication Union - Telecommunication Standardization Sector
JITC	Joint Interoperability Test Command
kbps	kilobits per second
Mbps	Megabits per second
MCS	Media Convergence Server
MFR1	Multi-Frequency Recommendation 1
min	minute
MLPP	Multi-Level Precedence and Preemption
MOS	Mean Opinion Score
NFAS	Non Facility Associated Signaling
NI 1/2	National ISDN Standard 1 or 2
NI2	National ISDN Standard 2
NIC	Network Interface Card
NX56	Data format restricted to multiples of 56 kbps
NX64	Data format restricted to multiples of 64 kbps
OAM	Operational Administration and Maintenance
PBX	Private Branch Exchange
PBX 1	Private Branch Exchange 1
PCM	Pulse Code Modulation
PCM-24	Pulse Code Modulation - 24 Channels
PCM-30	Pulse Code Modulation - 30 Channels
PRI	Primary Rate Interface
PSTN	Public Switched Telephone Network
Q.955.3	ISDN Signaling Standard for E1 MLPP
R	Required
RTCP	RTP Control Protocol
RTP	Real-time Transport Protocol
S/T	ISDN BRI 4-wire interface
SS7	Signaling System 7
STE	Secure Terminal Equipment
STIGs	Security Technical Implementation Guides
STU-III	Secure Telephone Unit -3rd generation
SUT	System Under Test
T1	Digital Transmission Link Level 1 (1.544 Mbps)
T1.619a	SS7 and ISDN MLPP Signaling Standard for T1
T.4	Standardization of Group 3 facsimile terminals for document transmission
TCP	Transmission Control Protocol
TFTP	Trivial File Transfer Protocol
UC	Unified Capabilities
UCR	Unified Capabilities Requirements
UDP	User Datagram Protocol
UPS	Uninterruptible Power Supply
VBD	Variable bit data
VoIP	Voice over Internet Protocol
VTC	Video Teleconferencing
yr	year